Administrator's Guide for
Avaya MultiVantage™ Software

Volumes 1, 2, and 3

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Notice

Every effort was made to ensure that the information in this document was complete and accurate at the time of printing. However, information is subject to change.

Preventing Toll Fraud

"Toll fraud" is the unauthorized use of your telecommunications system by an unauthorized party (for example, a person who is not a corporate employee, agent, subcontractor, or is not working on your company's behalf). Be aware that there may be a risk of toll fraud associated with your system and that, if toll fraud occurs, it can result in substantial additional charges for your telecommunications services.

Avaya Fraud Intervention

If you suspect that you are being victimized by toll fraud and you need technical assistance or support, in the United States and Canada, call the Technical Service Center's Toll Fraud Intervention Hotline at 1-800-643-2353.

How to Get Help

For additional support telephone numbers, go to the Avaya Web site:

http://www.avaya.com/support/

If you are:

- Within the United States, click *Escalation Lists*, which includes escalation phone numbers within the USA.
- Outside the United States, click *Escalation Lists* then click *Global Escalation List*, which includes phone numbers for the regional Centers of Excellence.

Providing Telecommunications Security

Telecommunications security (of voice, data, and/or video communications) is the prevention of any type of intrusion to (that is, either unauthorized or malicious access to or use of) your company's telecommunications equipment by some party.

Your company's "telecommunications equipment" includes both this Avaya product and any other voice/data/video equipment that could be accessed via this Avaya product (that is, "networked equipment").

An "outside party" is anyone who is not a corporate employee, agent, subcontractor, or is not working on your company's behalf. Whereas, a "malicious party" is anyone (including someone who may be otherwise authorized) who accesses your telecommunications equipment with either malicious or mischievous intent.

Such intrusions may be either to/through synchronous (time-multiplexed and/or circuit-based) or asynchronous (character-, message-, or packet-based) equipment or interfaces for reasons of:

- Utilization (of capabilities special to the accessed equipment)
- Theft (such as, of intellectual property, financial assets, or toll-facility access)
- Eavesdropping (privacy invasions to humans)
- Mischief (troubling, but apparently innocuous, tampering)
- Harm (such as harmful tampering, data loss or alteration, regardless of motive or intent)

Be aware that there may be a risk of unauthorized intrusions associated with your system and/or its networked equipment. Also realize that, if such an intrusion should occur, it could result in a variety of losses to your company (including but not limited to, human/data privacy, intellectual property, material assets, financial resources, labor costs, and/or legal costs).

Responsibility for Your Company's Telecommunications Security

The final responsibility for securing both this system and its networked equipment rests with you - Avaya's customer system administrator, your telecommunications peers, and your managers. Base the fulfillment of your responsibility on acquired knowledge and resources from a variety of sources including but not limited to:

- Installation documents
- System administration documents
- Security documents
- Hardware-/software-based security tools
- Shared information between you and your peers
- Telecommunications security experts

To prevent intrusions to your telecommunications equipment, you and your peers should carefully program and configure:

- Your Avaya-provided telecommunications systems and their interfaces
- Your Avaya-provided software applications, as well as their underlying hardware/software platforms and interfaces
- Any other equipment networked to your Avaya products.

Voice Over Internet Protocol (VoIP)

If the equipment supports Voice over Internet Protocol (VoIP) facilities, you may experience certain compromises in performance, reliability and security, even when the equipment performs as warranted. These compromises may become more acute if you fail to follow Avaya's recommendations for configuration, operation and use of the equipment. YOU ACKNOWLEDGE THAT YOU ARE AWARE OF THESE RISKS AND THAT YOU HAVE DETERMINED THEY ARE ACCEPTABLE FOR YOUR APPLICATION OF THE EQUIPMENT. YOU ALSO ACKNOWLEDGE THAT, UNLESS EXPRESSLY PROVIDED IN ANOTHER AGREEMENT, YOU ARE SOLELY RESPONSIBLE FOR (1) ENSURING THAT YOUR NETWORKS AND SYSTEMS ARE ADEQUATELY SECURED AGAINST UNAUTHORIZED INTRUSION AND (2) BACKING UP YOUR DATA AND FILES.

Standards Compliance

Avaya Inc. is not responsible for any radio or television interference caused by unauthorized modifications of this equipment or the substitution or attachment of connecting cables and equipment other than those specified by Avaya Inc. The correction of interference caused by such unauthorized modifications, substitution or attachment will be the responsibility of the user. Pursuant to Part 15 of the Federal Communications Commission (FCC) Rules, the user is cautioned that changes or modifications not expressly approved by Avaya Inc. could void the user's authority to operate this equipment.

The equipment described in this manual complies with standards of the following organizations and laws, as applicable:

- Australian Communications Agency (ACA)
- American National Standards Institute (ANSI)
- Canadian Standards Association (CSA)
- Committee for European Electrotechnical Standardization (CENELEC) – European Norms (EN's)
- Digital Private Network Signaling System (DPNSS)
- European Computer Manufacturers Association (ECMA)
- European Telecommunications Standards Institute (ETSI)
- FCC Rules Parts 15 and 68
- International Electrotechnical Commission (IEC)
- International Special Committee on Radio Interference (CISPR)
- International Telecommunications Union - Telephony (ITU-T)
- ISDN PBX Network Specification (IPNS)
- National ISDN-1
- National ISDN-2
- Underwriters Laboratories (UL)

Product Safety Standards

This product complies with and conforms to the following international Product Safety standards as applicable:

Safety of Information Technology Equipment, IEC 60950, 3rd Edition including all relevant national deviations as listed in Compliance with IEC for Electrical Equipment (IECEE) CB-96A.

Safety of Laser products, equipment classification and requirements:
- IEC 60825-1, 1.1 Edition
- Safety of Information Technology Equipment, CAN/CSA-C22.2 No. 60950-00 / UL 60950, 3rd Edition
- Safety Requirements for Customer Equipment, ACA Technical Standard (TS) 001 - 1997
- One or more of the following Mexican national standards, as applicable: NOM 001 SCFI 1993, NOM SCFI 016 1993, NOM 019 SCFI 1998

Electromagnetic Compatibility (EMC) Standards

This product complies with and conforms to the following international EMC standards and all relevant national deviations:

Limits and Methods of Measurement of Radio Interference of Information Technology Equipment, CISPR 22:1997 and EN55022:1998.

Information Technology Equipment – Immunity Characteristics – Limits and Methods of Measurement, CISPR 24:1997 and EN55024:1998, including:
- Electrostatic Discharge (ESD) IEC 61000-4-2
- Radiated Immunity IEC 61000-4-3
- Electrical Fast Transient IEC 61000-4-4
- Lightning Effects IEC 61000-4-5
- Conducted Immunity IEC 61000-4-6
- Mains Frequency Magnetic Field IEC 61000-4-8
- Voltage Dips and Variations IEC 61000-4-11
- Powerline Harmonics IEC 61000-3-2
- Voltage Fluctuations and Flicker IEC 61000-3-3

Federal Communications Commission Statement

Part 15:

Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Part 68: Answer-Supervision Signaling. Allowing this equipment to be operated in a manner that does not provide proper answer-supervision signaling is in violation of Part 68 rules. This equipment returns answer-supervision signals to the public switched network when:
- answered by the called station,
- answered by the attendant, or
- routed to a recorded announcement that can be administered by the customer premises equipment (CPE) user.

This equipment returns answer-supervision signals on all direct inward dialed (DID) calls forwarded back to the public switched telephone network. Permissible exceptions are:
- A call is unanswered.
- A busy tone is received.
- A reorder tone is received.

Avaya attests that this registered equipment is capable of providing users access to interstate providers of operator services through the use of access codes. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operator Consumers Act of 1990.

This equipment complies with Part 68 of the FCC Rules. On the rear of this equipment is a label that contains, among other information, the FCC registration number and ringer equivalence number (REN) for this equipment. If requested, this information must be provided to the telephone company.

The REN is used to determine the quantity of devices which may be connected to the telephone line. Excessive RENs on the telephone line may result in devices not ringing in response to an incoming call. In most, but not all areas, the sum of RENs should not exceed 5.0. To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company.

REN is not required for some types of analog or digital facilities.

Means of Connection

Connection of this equipment to the telephone network is shown in the following table.

Manufacturer's Port Identifier	FIC Code	SOC/REN/ A.S. Code	Network Jacks
Off/On premises station	OL13C	9.0F	RJ2GX, RJ21X, RJ11C
DID trunk	02RV2-T	0.0B	RJ2GX, RJ21X
CO trunk	02GS2	0.3A	RJ21X
CO trunk	02LS2	0.3A	RJ21X
Tie trunk	TL31M	9.0F	RJ2GX
Basic Rate Interface	02IS5	6.0F, 6.0Y	RJ49C
1.544 digital interface	04DU9-BN, 1KN, 1SN	6.0F	RJ48C, RJ48M
120A2 channel service unit	04DU9-DN	6.0Y	RJ48C

If the terminal equipment (for example, the MultiVantage™ Solution equipment) causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice is not practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

If trouble is experienced with this equipment, for repair or warranty information, please contact the Technical Service Center at 1-800-242-2121 or contact your local Avaya representative. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

It is recommended that repairs be performed by Avaya certified technicians.

The equipment cannot be used on public coin phone service provided by the telephone company. Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.

This equipment, if it uses a telephone receiver, is hearing aid compatible.

Canadian Department of Communications (DOC) Interference Information

This Class A digital apparatus complies with Canadian ICES-003. Cet appareil numérique de la classe A est conforme à la norme NMB-003 du Canada.

This digital apparatus does not exceed Class A limits for radio noise emission set out in the radio interference regulation of the Canadian Department of Communications.

Le Présent Appareil Nomérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils manicures de la class A préscrites dans le reglement sur le brouillage radioélectrique édicté par le ministére des Communications du Canada.

This equipment meets the applicable Industry Canada Terminal Equipment Technical Specifications. This is confirmed by the registration number. The abbreviation, IC, before the registration number signifies that registration was performed based on a Declaration of Conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

DECLARATIONS OF CONFORMITY

United States FCC Part 68 Supplier's Declaration of Conformity (SDoC)

Avaya Inc. in the United States of America hereby certifies that the equipment described in this document and bearing a TIA TSB-168 label identification number complies with the FCC's Rules and Regulations 47 CFR Part 68, and the Administrative Council on Terminal Attachments (ACTA) adopted technical criteria.

Avaya further asserts that Avaya handset-equipped terminal equipment described in this document complies with Paragraph 68.316 of the FCC Rules and Regulations defining Hearing Aid Compatibility and is deemed compatible with hearing aids.

Copies of SDoCs signed by the Responsible Party in the U. S. can be obtained by contacting your local sales representative and are available on the following Web site:

http://support.avaya.com/elmodocs2/DoC/SDoC/index.jhtml/

All MultiVantage™ system products are compliant with FCC Part 68, but many have been registered with the FCC before the SDoC process was available. A list of all Avaya registered products may be found at:

http://www.part68.org/

by conducting a search using "Avaya" as manufacturer.

European Union Declarations of Conformity

Avaya Inc. declares that the equipment specified in this document bearing the "CE" (*Conformité Europeénne*) mark conforms to the European Union Radio and Telecommunications Terminal Equipment Directive (1999/5/EC), including the Electromagnetic Compatibility Directive (89/336/EEC) and Low Voltage Directive (73/23/EEC). This equipment has been certified to meet CTR3 Basic Rate Interface (BRI) and CTR4 Primary Rate Interface (PRI) and subsets thereof in CTR12 and CTR13, as applicable.

Copies of these Declarations of Conformity (DoCs) signed by the Vice President of MultiVantage™ Solutions research and development, Avaya Inc., can be obtained by contacting your local sales representative and are available on the following Web site:

http://support.avaya.com/elmodocs2/DoC/IDoC/index.jhtml/

Japan

This is a Class A product based on the standard of the Voluntary Control Council for Interference by Information Technology Equipment (VCCI). If this equipment is used in a domestic environment, radio disturbance may occur, in which case, the user may be required to take corrective actions.

この装置は、情報処理装置等電波障害自主規制協議会（ＶＣＣＩ）の基準に基づくクラスＡ情報技術装置です。この装置を家庭環境で使用すると電波妨害を引き起こすことがあります。この場合には使用者が適切な対策を講ずるよう要求されることがあります。

Network Connections

Digital Connections - The equipment described in this document can be connected to the network digital interfaces throughout the European Union.

Analogue Connections - The equipment described in this document can be connected to the network analogue interfaces throughout the following member states:

Belgium	Germany	Luxembourg
Netherlands	Spain	United Kingdom

LASER Product

The equipment described in this document may contain Class 1 LASER Device(s) if single-mode fiber-optic cable is connected to a remote expansion port network (EPN). The LASER devices operate within the following parameters:

- Maximum power output –5 dBm to -8 dBm
- Center Wavelength 1310 nm to 1360 nm
- CLASS 1 LASER PRODUCT IEC 60825-1: 1998

Use of controls or adjustments or performance of procedures other than those specified herein may result in hazardous radiation exposure. Contact your Avaya representative for more laser product information.

To order copies of this and other documents:

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Attention: Avaya Account Management
E-mail: totalware@gwsmail.com

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Volume 2

Volume 3

About this document

Overview

This document describes the May release of Avaya MultiVantage administration, and includes all incremental releases up to and including Release 11. You may also want to see the *Avaya Multivantage running on aDEFINITY Server Change Description* to see what is new.

MultiVantage is a family of cost-effective digital communication systems. These systems:

- route voice and data information between various endpoints (telephones, terminals, computers)
- provide highly robust networking capabilities
- include an extensive set of standard features including attendant consoles, voice processing interface, call coverage, DS1 (T1 and E1) connectivity, hospitality support, recorded announcements, and trunk-to-trunk transfer
- provide flexibility and allow for the addition of optional features and/or upgrades to the system as business needs change

Purpose

This document provides an overall reference for planning, operating, and administering your MultiVantage solution. The book is divided into three volumes that present information on how to perform administrative tasks, how to complete administrative screens, and more detailed information on individual features.

This document does not contain information about how to install, maintain, repair, or troubleshoot the switch.

Audience

This document is intended for MultiVantage system administrators and managers, users interested in information about specific features, and Avaya personnel responsible for planning, designing, configuring, selling, and supporting the system.

How to use this document

Use this document as a guide to completing administrative procedures and as a reference document. If you are interested in information about a particular task, screen, or feature, use the index or table of contents to locate the page number where the information is described.

Organization

The first volume of this document provides step-by-step tasks for the administrative procedures that implement MultiVantage features. The second volume explains how to fill out MultiVantage screens and defines the values for the fields on the screens. The third volume explains MultiVantage features and provides additional detail about switch operations.

The following list describes the sections in this book.

''System basics'' tells you how to log in and log off, set permissions for others who use the administration terminal, set daylight savings rules, set the system date and time, post messages, and back up the information you administer.

''Introduction to the MultiVantage system'' provides information on system-wide functions. It explains how to read, use, and make simple changes to your dial plan, and how to assign feature access codes.

''Managing phones'' explains how to add, swap, and remove phones on your system, and how to customize a phone for a switch administrator.

''Managing phone features'' explains how to administer feature buttons for your users' phones.

''Managing your attendant consoles'' explains attendant console feature buttons, and tells you how to change, move, or add attendant consoles.

''Managing displays'' provides information on the messages that appear on the read-out screen on display phones.

‘‘Handling incoming calls’’ shows you how to set up call coverage for incoming calls to be sure that incoming calls are answered when the called party is not available.

‘‘Routing outgoing calls’’ explains how the switch handles outgoing calls and tells you how to modify call restrictions and your routing plan.

‘‘Managing multimedia calling’’ describes the Multimedia Applications Server Interface (MASI), and provides instructions on administration, monitoring, and troubleshooting. This section also provides information on Multimedia Call Handling (MMCH), which enables users to control voice, video, and data transmissions using a telephone and PC.

‘‘Setting up telecommuting’’ provides information on switch-wide settings and individual administration for telecommuting.

‘‘Enhancing system security’’ provides information on analyzing and setting up basic system security, preventing toll fraud, using logins and permissions and passwords, and dealing with security violations.

‘‘Managing trunks’’ contains procedures for working with analog and digital trunks. Specialized trunks such as APLT, tandem, release-link, DMI-BOS and ISDN trunk groups are not covered in this manual.

‘‘Managing announcements’’ tells you how to record, save, copy, restore and delete announcements.

‘‘Managing group communication’’ shows you how to administer your system so users can page other users or use their phones as intercoms. You can also give specific users permission to monitor other users’ calls or to interrupt active calls with important messages.

‘‘Managing data calls’’ describes the system features available to enable data communications.

‘‘Collecting billing information’’ provides information on account codes, and on tracking and collecting information about calls.

‘‘Screen reference’’ provides a brief description and a graphic representation of the screens used for MultiVantage administration. It also lists the valid values for the fields on the screens, and describes when and why to use each value.

‘‘Command reference’’ Use the commands in these tables to access each administration screen.

‘‘Phone reference’’ describes many of the telephones that you can connect to the MultiVantage Solution. It also describes the unique features and buttons for each phone series to help you administer your user phones.

‘‘Features and technical reference’’ is a comprehensive technical reference for feature information.

Task-related information

The information for each task is usually presented under the following headings:

- Task

  Identifies the administrative procedure and gives a brief explanation of what is accomplished by completing the task.

- Before you start

  Lists hardware that must be installed or other tasks that must be completed before starting the task.

- Instructions

  Begins with a short introduction to set up an example, then provides step-by step, numbered instructions on how to complete the administrative task. Screen pictures and background or decision-making information are provided when appropriate.

- Fixing problems

  This section is not included in all task sections. It provides a brief coverage of possible problems, possible causes, and suggested solutions.

- More information

  Presents additional technical information that pertains directly to the completion of the current task.

- Related topics

  Provides cross-references to related tasks or related feature references.

Feature-related information

The information for each feature is usually presented under four headings:

- Feature title

  Gives the name and a brief overview of the feature. Tells what it does or how it serves the system.

- Detailed description

  Provides more detailed, technical information about a feature. When appropriate, additional guidelines and examples are provided. In some cases, expanded technical information is provided on one or several aspects of the feature.

- Interactions

  Lists and briefly discusses other features that may significantly affect a feature.

- Related topics

  Provides cross-references to related tasks, features, or screens.

Conventions used in this document

Become familiar with the following terms and conventions. They help you use this book with your MultiVantage system.

- To "move" to a certain field, you can use the TAB key, arrows, or the RETURN key.
- A "screen" is a screen form displayed on the terminal monitor.
- In this book we use the terms "telephone" and "voice terminal" to refer to phones.
- If you use terminal emulation software, you need to determine which keys correspond to ENTER, RETURN, CANCEL, HELP, NEXT PAGE, etc.
- Commands are printed in bold face as follows: **command**.
- Keys and buttons are printed as follows: KEY.
- Screen displays are printed in constant width as follows: `screen display`.
- Variables are printed in italics as follows: *variable*.
- We show complete commands in this book, but you can always use an abbreviated version of the command. For example, **list configuration station** can be entered as **list config sta**.

- We show commands and screens from the newest MultiVantage system and refer to the most current books. Please substitute the appropriate commands for your system and refer to the manuals you have available.
- If you need help constructing a command or completing a field entry, remember to use HELP.
- When you press HELP at any point on the command line, a list of available commands appears.
- When you press HELP with your cursor in a field on a screen, a list of valid entries for that field appears.
- The status line or message line can be found near the bottom of your monitor display. This is where the system displays messages for you. Check the message line to see how the system responds to your input. Write down the message if you need to call our helpline.
- When a procedure requires you to press ENTER to save your changes, the screen you were working on clears and the cursor returns to the command prompt. The message line shows "`command successfully completed`" to indicate that the system accepted your changes.

Tip:
Draws attention to information that you may find helpful.

NOTE:
Draws attention to information that you must heed.

CAUTION:
Denotes possible harm to software, possible loss of data, or possible service interruptions.

WARNING:
Denotes possible harm to hardware or equipment.

SECURITY ALERT:
Indicates when system administration may leave your system open to toll fraud.

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How to get this book on the web

If you have internet access, you can view and download the latest version of the *Administrator's Guide for Avaya MultiVantage™ Software*. To view the book, you must have a copy of Acrobat Reader.

To access the latest version:

1. Access the Avaya web site at http://avaya.com
2. Click **Support**.
3. Click **Online Services**.
4. Click **Documentation**.
5. Click **Recent Documents**.
6. Scroll down to find the latest release of MultiVantage documents.
7. Search for **555-233-506** (the document number) to view the latest version of the book.

How to get help

If you need additional help, the following resources are available. You may need to purchase an extended service agreement to use some of these resources. See your Avaya representative for more information.

■ DEFINITY Helpline (for help with feature administration and system applications)	+1-800-225-7585
■ Avaya National Customer Care Center Support Line (for help with maintenance and repair)	+1-800-242-2121
■ Avaya Toll Fraud Intervention	+1-800-643-2353
■ Avaya Corporate Security	+1-800-822-9009
	+1-925-224-3401
■ Avaya Centers of Excellence	
— North America	1-800-248-1111
— Central/Latin America, Caribbean (for dealers only)	Contact your local representative
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— Japan	+813-5575-8800
— Shanghai	+8621-5459-4590
— Singapore	+65-872-8686
— Canada	1-800-387-4268

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System basics

1

This section provides the basic step-by-step procedures you need to manage your Avaya MultiVantage solution. It explains how to log in and log off, set permissions for others who use the administration terminal, set daylight savings rules, set date and time, post messages, and back up the information you administer.

Logging into the system

You must log in before you can administer your system. If you are performing remote administration, you must establish a remote administration link and possibly assign the remote administration extension to a hunt group before you log in. The members of this hunt group are the extensions of the data modules available to connect to the system administration terminal. For information about setting up remote administration, contact your Avaya representative.

NOTE:
Change your password frequently, at least once a month, to help keep hackers out of your system. For instructions on how to change your password, refer to ''Changing passwords'' on page 393.

When not using the system, log off for security purposes.

Instructions

Logging into the system

This procedure provides instructions for logging in from the system terminal not a remote terminal.

To log into the system:

1. Enter your login name and press RETURN.
2. Enter your password and press RETURN.

   For security, your password does not display as you type it.
3. Enter the kind of terminal you have or the type your system emulates and press RETURN.

   The Command prompt appears.

   NOTE:
   If you enter the wrong terminal type, it can lock up your system. If the system is not responding to keyboard commands, type **newterm** and press RETURN. Enter the correct terminal type on the new screen and press RETURN. If this does not work, turn the power off only on the terminal and then turn it back on. The terminal reboots and you can login again.

Logging in for remote administration

To log in for remote administration:

1. Dial the Uniform Call Distribution (UCD) group extension number.

   NOTE:
   The UCD group extension number is assigned when you set up remote administration.

   - If you are off-premises, use the Direct Inward Dialing (DID) number, an Listed Directory Number (LDN) (you must use a phone), or the trunk number dedicated to remote administration.

- If you are on-premises, use an extension number.

  If you dialed a DID number, dedicated trunk number, or extension, you receive data tone or visually receive answer confirmation.

  If an LDN was dialed, the attendant will answer.

  a. Ask to be transferred to the UCD group extension number.

     You receive data tone or visually receive answer confirmation.

  b. Transfer the voice call to your data terminal.

  The Login prompt appears.

2. Complete the steps for ''Logging into the system'' on page 27.

   For information about setting up remote administration, contact your Avaya representative.

Failed log-ins

You can administer your system for a maximum number of failed login attempts. For instance, if you set the maximum limit to 4 and there are 4 login failures in 10 minutes or less, the login will be locked out for 5 minutes. An account lockout event is logged to the security log.

Accessing the Avaya S8700 Media Server

To administer the Avaya™ S8700 Media Server, you must be able to access it. Personal computers and services laptop computers equipped with a network PCMCIA card and Avaya Site Administration (ASA) and a Web browser are the primary support access for system initialization, aftermarket additions, and continuing maintenance.

You can access the Avaya S8700 Media Server in one of three ways:

- directly
- remotely over the customer's local area network (LAN)
- over a modem

A direct connection and over the customer's LAN are the preferred methods. Remote access over a modem is for Avaya maintenance access only.

Accessing the Avaya S8700 Media Server Directly

You can access the S8700 Media Server directly by plugging a computer into the services port which defaults to port 2 (Eth1) on the back of the media server. You must use a crossover cable with an RJ45 connector on each end. Plug the other end into the network connector (NIC card) on the your computer. You may need a NIC card adapter.

Once connected, you can administer the media server using three tools:

- Media server web interface for server-specific administration.
- ASA for Avaya MultiVantage software features
- A command line interface in a Telnet window for Avaya MultiVantage software features.

Media Server Web Interface

You can access the media server web interface either by connecting directly to the services port on the media server or connecting over the customoer's network.

Connected to the services port

To use the media server web interface:

1. Open either the Netscape or MS Internet Explorer browser.
2. In the Location/Address field, type **192.11.13.6** and press ENTER.
3. When prompted, log in to administer the S8700 Media Server and the Avaya MultiVantage™ software features.

Connected to the customer network

To use the media server web interface:

1. Open either the Netscape or MS Internet Explorer browser.
2. In the Location/Address field, type the **active server** name or IP address and press ENTER.
3. When prompted, log in to administer the S8700 Media Server and the Avaya MultiVantage™ software features.

   You may also connect directly to an individual server using its name or IP address.

Avaya Site Administration

To use ASA:

1. You must have a PC or laptop with Avaya™ Site Administration on it.
2. You must be connected over a LAN to either:
   - a dedicated LAN port for service technicians, or
   - a customer LAN port

Instructions

1. Start up ASA by double-clicking on the ASA icon.
2. In the Target System field, use the pull-down menu to select the desired system.

   If you have not set up the desired switch, click on the NEW SWITCH icon (on the left of the toolbar row). Complete the Add Switch wizard to build the system record. When completed, the switch record will be available in the Target System pull-down list.
3. Click on Start GEDI.

   You now are connected to the desired system.

Accessing the S8700 Media Server Remotely over the network

You can access the S8700 Media Server from any computer connected through the LAN. To access either server, use the IP address assigned to the server you want to access. You can also use the **active server** address to connect automatically to the server that is active. Once connected, you can administer the media server using three tools:

- Media server web interface for server-specific administration and call processing features
- Avaya Site Administration for the MultiVantage software (Only available on the active server)
- A command line interface via a Telnet window for Linux shell commands

Installing Avaya Site Administration

If you do not have ASA on your computer, make sure your PC or laptop first meets the minimum requirements.

Table 1. Minimum requirements to run ASA

Operating systems MS Windows 95 MS Windows 98 MS Windows NT 4.0 MS Windows 2000	**Processor/RAM** 486SL/66/16 MB Pentium/32 MB Pentium/64 MB Pentium-class 300 MHz/64 MB
Graphics adapter	SVGA with minimum screen resolution of 800 x 600
Floppy disk drive	3-1/2 in. 1.44-MB floppy disk drive
CD-ROM	CD-ROM drive (required to install ASA from CD)
Available hard disk space	A minimum of 100-MB free hard disk space is required. The requirement for disk space depends on the size and number of configuration data sets.
Printer port	Standard PC or network printer port is required for connecting to a printer to produce station button labels.
Network adapter	Required for network access to the Avaya S8700 Media Server, AUDIX, and other network-connected systems.
Free serial ports	One free serial port capable of 9600-bps operation is required for a connection to each serial device (UPS). We recommend that PCs have at least a 16550A UART or 16550A UART simulator (capable of 56 kbps DTE-speed connections). USB and internal modems should emulate this hardware. A second serial port is required for simultaneous connection to AUDIX through a serial connection.

You can install ASA on your computer two different ways: from the Avaya Site Administration CD or from the S8700 Media Server.

From the ASA CD

Place the ASA CD in the CD-ROM drive and follow the installation instructions in the install wizard.

From the S8700 Media Server

1. Connect to one of the media servers either directly or through the network from the computer you want ASA to be installed on.
2. Access the media server web interface by typing in the Location/Address field 192.11.13.6 (direct connect) or the IP address or the domain name of the media server (over the network).
3. Log in and get to the main menu.
4. Click on Download ASA to download Avaya Site Administration to your computer.
5. Follow the installation instructions in the install wizard.

To start ASA, click on

Start > Programs > Avaya Site Administration.

ASA supports a terminal emulation mode, which is directly equivalent to using SAT commands on a dumb terminal or through a Telnet session. ASA also supports a whole range of other features, including the graphically enhanced MultiVantage interface (GEDI) and Data Import. For more information refer to the Help, Guided Tour, and Show Me accessed from the ASA Help menu.

Logging in with Access Security Gateway

Access Security Gateway (ASG) is an authentication interface used to protect the system administration and maintenance ports and logins on Avaya MultiVantage solution. ASG uses a challenge and response protocol to validate the user and reduce unauthorized access.

You can administer ASG authentication on either a port type or login ID. If you set ASG authentication for a specific port, it restricts access to that port for all logins. If you set ASG authentication for a specific login ID, it restricts access to that login, even when the port is not administered to support ASG.

Authentication is successful only when Avaya MultiVantage and ASG communicate with a compatible key. You must maintain consistency between the Access Security Gateway Key and the secret key assigned to the Avaya MultiVantage login. For more information about ASG, refer to ''Using access security gateway'' on page 387.

Before you start

Before you can log into the system with ASG authentication, you need an Access Security Gateway Key, and you need to know your personal identification number (PIN). The Access Security Gateway Key must be pre-programmed with the same secret key (such as, ASG Key, ASG Passkey, or ASG Mobile) assigned to the Avaya MultiVantage login.

Verify that the Access Security Gateway (ASG) field on the System-Parameters Customer Options screen is set to **y**. If not, contact your Avaya representative.

Instructions

To log into the system with ASG:

1. Enter your login ID and press RETURN.

   The system displays the challenge number (for example, 555-1234) and system Product ID number (for example, 1000000000). The Product ID provides Avaya Services with the specific Avaya MultiVantage system identifier.

2. Press ON to turn on your Access Security Gateway Key.

3. Type your PIN and press ON.

   The Access Security Gateway Key displays a challenge prompt.

4. At the challenge prompt on the Access Security Gateway Key, type the challenge number without the "-" character (for example, 5551234) from your screen and press ON.

   The Access Security Gateway Key displays a response number (for example, 999-1234).

5. At the response prompt on your terminal, type the ASG response number without the "-" character (for example, 9991234) and press RETURN.

   The Command prompt appears.

   NOTE:
   If you make 3 invalid login attempts, the system terminates the session. For more information, refer to the appropriate maintenance book for your system.

Fixing problems

When logging in failures occur, if you are a super-user, you can use the list asg-history command to determine the cause. The asg history log contains the last 100 or 250 records depending on your system. This log contains the date and time, the port mnemonic, the login ID entered (correct or incorrect), and the status for each session attempt. For specific information about the ASG history log, refer to *Reports for Avaya MultiVantage Software*.

Logging off the system

For security, log off any time you leave your terminal. If you use terminal emulation software to administer the switch, log off the system and exit the emulation application before switching to another software package.

Instructions

To log off:

1. Type **logoff** and press RETURN.

   If the Facility Test Calls or Remote Access features are administered, Alarm origination is disabled, or if you have busied out resources or active minor or major alarms, a security screen appears. You may want to take appropriate action (for example, disable these features or address any alarms) before you log off.

   If none of the above special circumstances exist, the system logs you off.

```
Facility Test Call Administered
Remote Access Administered
Alarm Origination is currently disabled
Active major/minor alarm detected; be sure to resolve it

Proceed with Logoff? [n]
```

2. At the `Proceed with Logoff` prompt, type **y** to log off.

   If you log off with alarm origination disabled and the system generates an alarm, Avaya support services will not receive any notification of the alarm. For more information about alarms, refer to the maintenance book for your system.

Setting command permissions

Avaya MultiVantage allows you to modify the permissions associated with a login. The system maintains default permissions for each level of login, but you may want to further restrict the login, or at least make sure the defaults are appropriate for the user. The default values for these fields vary based on the login type.

When set to **y**, the permissions on the Command Permission Categories screen apply for any object that is not restricted. The second and third pages of the Command Permission Categories screen allow you to restrict the user from any access to specified objects. If you want to limit a user's permissions beyond those on page one, enter the objects in this list. For example, if you want a user to be able to add and change stations, but not VDNs, you can enter **y** in the Administer Stations field and the Additional Restrictions field. Then on page 2 or 3, enter vdn as a restricted object.

Instructions

In our example, we set the permissions necessary to allow the user to administer daylight savings time rules.

To change command permissions:

1. Type **change permissions sup3ru** and press RETURN.

   The Command Permission Categories screen appears.

```
                        COMMAND PERMISSION CATEGORIES
                          Login Name: sup3ru
COMMON COMMANDS
               Display Admin. and Maint. Data? y
                          System Measurements? y

ADMINISTRATION COMMANDS
      Administer Stations? y           Administer Features? y
        Administer Trunks? y        Administer Permissions? y
  Additional Restrictions? n

MAINTENANCE COMMANDS
        Maintain Stations? n      Maintain Switch Circuit Packs? n
          Maintain Trunks? n     Maintain Process Circuit Packs? n
         Maintain Systems? n              Maintain Enhanced DS1? n
```

2. Type **y** in the Display Admin and Maint Data field.
3. Type **y** in the Administer Features field.
4. Press ENTER to save your work.

More information

There are 2 types of users — superuser and non-superuser.

- A superuser provides access to the add, change, display, list, and remove commands for all customer logins and passwords. The superuser can administer any mix of superuser/nonsuperuser logins. The superuser can administer between 10 and 19 logins depending on your system.

  The Avaya™ S8100 Media Server with CMC1 Media Gateway system allows up to 14 simultaneous connections (logins) (MultiVantage can have 5 connections, AUDIX can have 4 connections, and the rest of the connections are reserved for shell commands.)

  Logins must be 3 to 6 alphabetic/numeric characters, or a combination of both.

  NOTE:
  If several users are logging in and out at the same time, a user may see the message: `Transient command conflict detected; please try later`. After the "users" have completed logging in or out, the terminal is available for use.

- A nonsuperuser may change their password with permission set by the superuser. However, once a password has been changed, the nonsuperuser must wait 24 hours before changing the password again.

Establishing daylight savings rules

Avaya MultiVantage allows you to set the daylight savings time rules so features, such as time-of-day routing and call detail recording (CDR), adjust automatically to daylight savings time. The correct date and time assure that CDR records are correct. You can set daylight savings time rules to transition to and from daylight savings time outside of normal business hours, so the number of affected CDR records is small.

You can set up 15 customized daylight savings time rules. This allows administrators with switches in several different time zones to set up a rule for each. A daylight savings time rule specifies the exact time when you want to transition to and from daylight savings time. It also specifies the increment at which to transition (for example, 1 hour).

Instructions

Establishing daylight savings rules

In our example, we set daylight savings time rules.

To modify a daylight savings rule:

1. Type **change daylight-savings-rules** and press RETURN.

   The Daylight Savings Rules screen appears.

```
                              DAYLIGHT SAVINGS RULES
Rule           Change Day                      Month     Date    Time     Increment

0:   No Daylight Savings

1:     Start: first Sunday___  on or after April___   1  at  _2:00  01:00
       Stop:  first Sunday___  on or after October_25  at  _2:00
2:     Start: first _________  on or after ________ __  at  __:__
       Stop:  first _________  on or after ________ __  at  __:__
3:     Start: first _________  on or after ________ __  at  __:__
       Stop:  first _________  on or after ________ __  at  __:__
4:     Start: first _________  on or after ________ __  at  __:__
       Stop:  first _________  on or after ________ __  at  __:__
5:     Start: first _________  on or after ________ __  at  __:__
       Stop:  first _________  on or after ________ __  at  __:__
6:     Start: first _________  on or after ________ __  at  __:__
       Stop:  first _________  on or after ________ __  at  __:__
7:     Start: first _________  on or after ________ __  at  __:__
       Stop:  first _________  on or after ________ __  at  __:__
```

2. Complete the Start fields for rule 1.

   Rule 1 applies to all time zones in the U.S. and begins on the first Sunday on or after 1 April at 2:00 a.m. with a 01:00 increment. Daylight Savings Time stops on the first Sunday on or after 25 October at 2:00 a.m. also with a 01:00 increment (used as a decrement when switching back to Standard time.

   The increment is added to standard time at the specified start time and the clock time shifts by that increment (for example, for 01:59:00 to 01:59:59 the clock time shows 01:59 and at 02:00 the clock shows 03:00).

   On the stop date, the increment is subtracted from the specified stop time (for example, for 01:59:00 to 01:59:59 the clock time shows 01:59 and at 02:00 the clock shows 01:00).

   a. Type **Sunday** in the Change Day field.

   b. Type **April** in the Month field.

   c. Type **1** in the Date field.

   d. Type **2:00** in the Time field.

e. Type **1:00** in the Increment field.

This information specifies the day, month, date, and time and increment at which you want the system clock to transition to daylight saving time.

NOTE:
You cannot delete a daylight savings rule if it is in use on either the Locations or Date and Time screens. However, you can change any rule except rule 0 (zero).

3. Complete the Stop fields for rule 1.

   a. Type **Sunday** in the Change Day field.

   b. Type **October** in the Month field.

   c. Type **25** in the Date field.

   d. Type **2:00** in the Time field.

   This information specifies the day, month, date, and time you want the system clock to transition back to standard time.

4. Press ENTER to save your changes.

Displaying daylight savings time rules

To display daylight savings time rules:

1. Type **display daylight-savings-rules** and press RETURN.

   The Daylight Savings Rule screen appears. Verify the information you entered is correct.

   Setting the system date and time

Update the date and time for events such as a leap year, the change to or from daylight savings time, or a system restart after a power failure. The correct date and time assure that CDR records are correct. CDR does not work until the date and time have been entered.

NOTE:
Changing the date and time may modify CDR data by 9 hours and 59 minutes. Therefore, you should change the date and time after normal business hours. After you change the date and time, review the time settings for any adjunct (other than AUDIX) linked to your system that uses the system time.

Before you start

Before you can set the date and time, you need to know whether it is currently daylight savings or standard time and know which daylight savings rule number you want to use. Daylight savings rule numbers are located on the Daylight Savings Rule screen.

Setting the system date and time

In our example, we set the date and time to Tuesday, November 3 at 8:30 p.m. standard time.

To set the system date and time:

1. Type **set time** and press RETURN.

   The Date and Time screen appears.

```
                              DATE AND TIME
DATE
      Day of the Week: Tuesday      Month: November
     Day of the Month: 3             Year: 1998

TIME
     Hour: 20    Minute: 30     Second: XX     Type: standard
               Daylight Savings Rule: 1
```

2. Complete the Date fields.
   a. Type **Monday** in the Day of the Week field.
   b. Type **November** in the Month field.
   c. Type **3** in the Day of the Month field.
   d. Type **1998** in the Year field.
3. Complete the Time fields.

   Use the 24-hour clock to set the hour, so if the current time is 2:00 p.m., you enter **14:00**. You cannot update `Second` — it automatically resets to 0 when you save your changes.
   a. Type **20** in the Hour field.
   b. Type **30** in the Minute field (8:30 p.m.).
   c. Type **standard** in the Type field.
   d. Type **1** in the Daylight Savings Rule field.

4. Press ENTER to save your changes.

NOTE:
When you change the date or time, some display phones may not automatically refresh the display. If this occurs, have each user press the date/time button on their phone to update the display.

Displaying the system date and time

To display the system date and time:

1. Type **display time** and press RETURN.

   The Date and Time screen appears. Verify the information you entered is correct.

Related topics

Refer to ''Establishing daylight savings rules'' for more information about setting system time.

For additional information, see *MultiVantage Guide to ACD Call Centers*.

Setting time of day clock synchronization

Time of Day Clock Synchronization enables a switch to synchronize its internal clock to UTC time provided by Internet time servers. Avaya uses the platform (LINUX or Windows) system clock connected to an Internet time server to provide time synchronization. The interface for these systems is web-based for LINUX and Windows, or command line, for Windows only.

LINUX is used in:

- Avaya™ S8700 Media Server for IP Connect Configurations
- Avaya™ S8700 Media Server for Multi-Connect Configurations
- Avaya™ S8700 Media Server with Avaya™ G700 Media Gateway

Windows is used in:

- Avaya™ S8100 Media Server with CMC1 Media Gateway
- Avaya™ S8100 Media Server with Avaya™ G600 Media Gateway

For additional information, see *Avaya MultiVantage Guide to ACD Call Centers*.

The following systems use Avaya™ Site Administration (ASA) for time synchronization.

- DEFINITY Server R
- DEFINITY Server CSI
- DEFINITY Server SI

Before you start

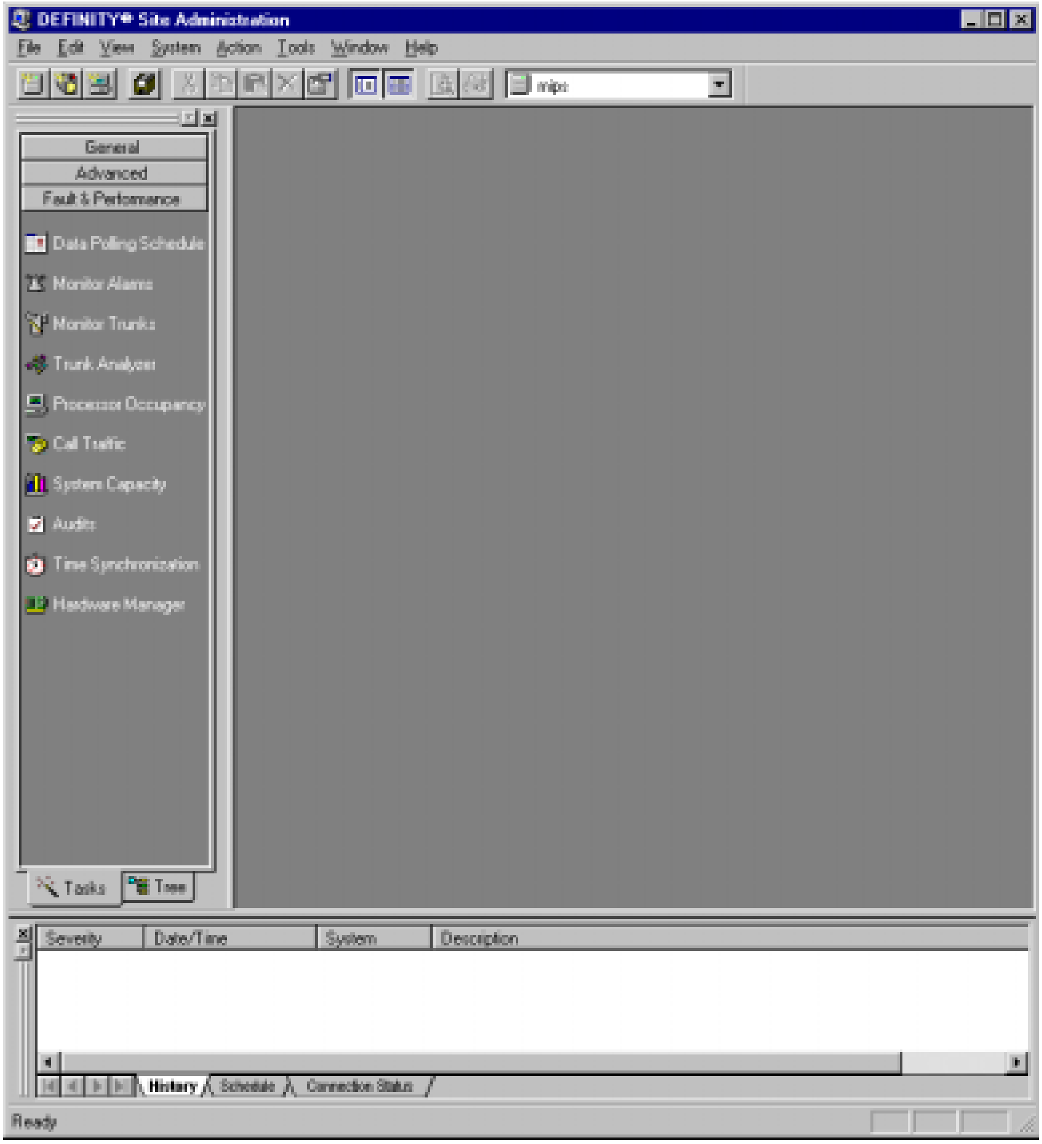
- A standard TCP/IP LAN connection is required to connect to the Internet time servers. If a LAN connection is not available, time sync will be done by setting the platform clock manually through the command line or web interface.
- You can set the following NTP/SNTP configuration information.
- On the target switch, verify if Daylight Savings Time is on.

Instructions

NOTE:
These instructions are for the DEFINITY Server R, DEFINITY Server CSI, and DEFINITY Server SI systems. For additional information, see *Avaya MultiVantage Guide to ACD Call Centers*.

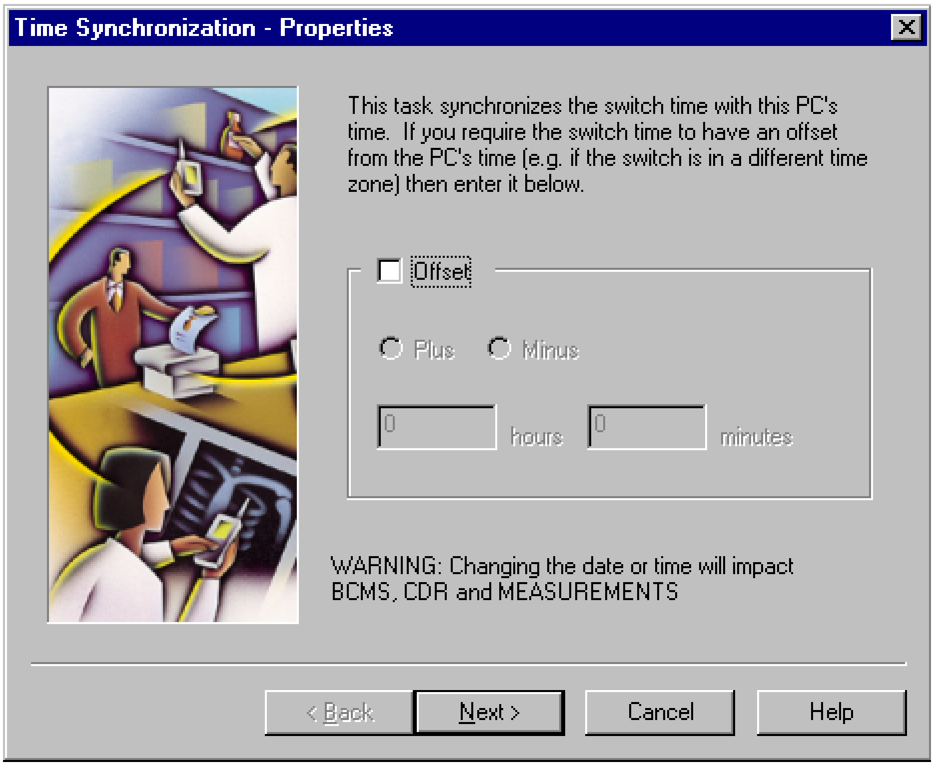
1. Activate the Avaya Site Administration screen.
2. Click on the Fault & Performance tab.

The Fault & Performance icons display.



3. Click on Time Synchronization.

   The Time Synchronization - Properties screen displays.

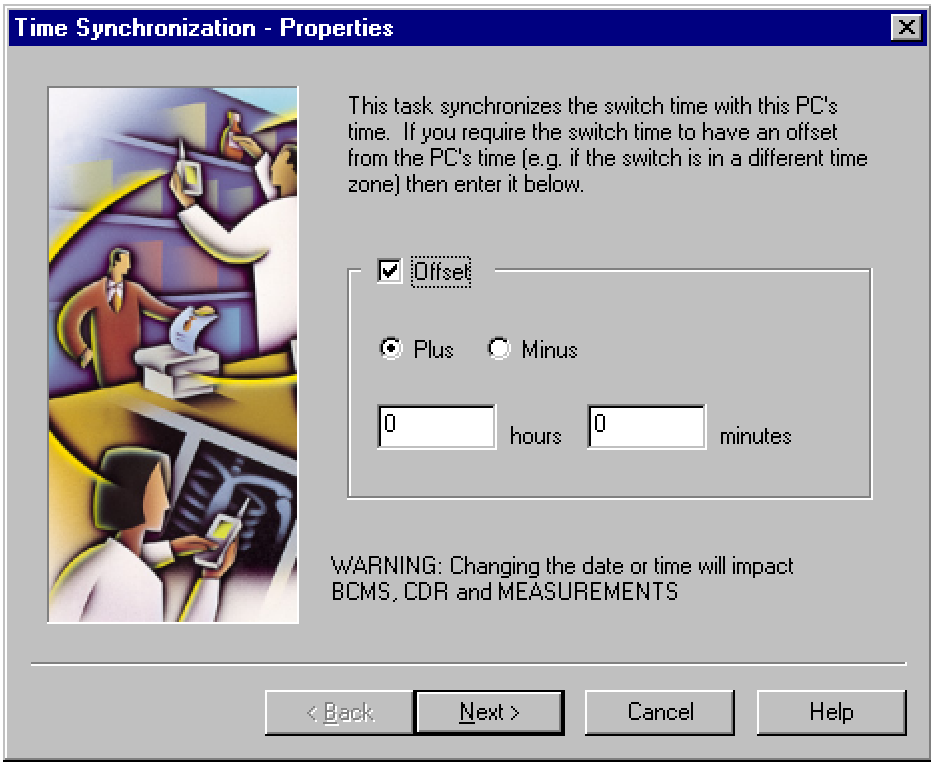


4. Click in the "Offset" box.

   The Plus and Minus radio buttons and the Hours and Minutes fields display.

5. Click NEXT.

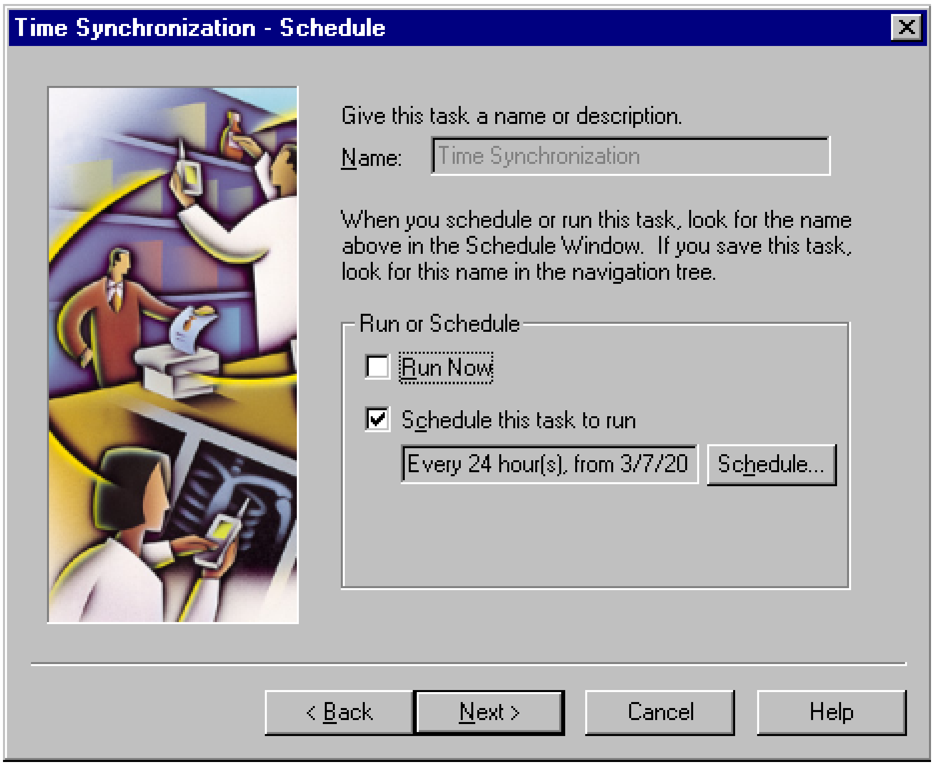
   The Time Synchronization - Properties screen displays.



6. Click Plus to add hours to the remote station (located to the west of the system time) or click Minus to subtract hours to the remote station (located to the east of the system time).
7. In the hours field, enter the number of hours to be added or subtracted to synchronize with the remote site.

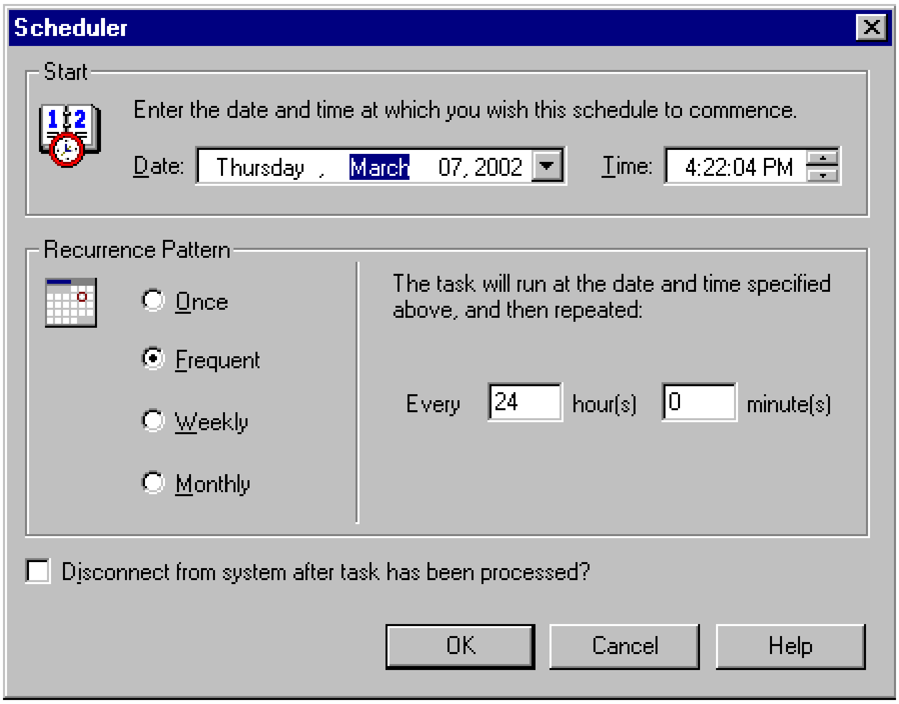
8. Click NEXT.

   The Time Synchronization - Schedule displays.



9. Select either:
   - "Run Now" to run this program immediately and click NEXT.
   - "Schedule this task to run" and check the field below to determine if the default setting is satisfactory. If this setting is not satisfactory, click SCHEDULE.

   The Scheduler screen displays.

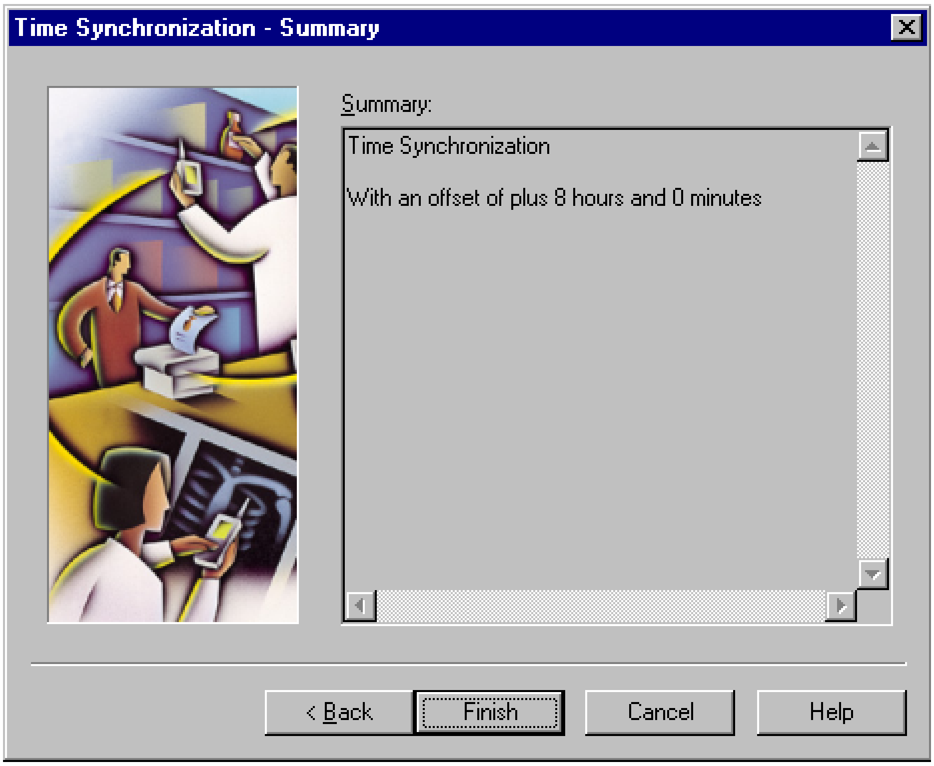


10. In the Date field, highlight each object and use the pull-down menu to select the desired setting.
11. In the Time field, highlight each item of time and use the pull-down menu to select the desired time.
12. In the Recurrent Pattern area, select one radio button for the desired frequency. Notice that each radio button is explained in the area to the immediate right.
13. If there are times to be included, set to the desired frequency.
14. Click OK.

    The Time Synchronization - Schedule displays.

15. Click NEXT.

    The Time Synchronization - Summary displays.



16. If the time synchronization description is satisfactory, click FINISH.

    If the time synchronization is not satisfactory, click BACK and revise the necessary information.

Related topics

Refer to *Avaya MultiVantage Call Center Software Guide to ACD Call Centers, 555-230-716*, for more information about using time of day clock synchronization.

Using the bulletin board

Avaya MultiVantage allows you to post information to the bulletin board. You can also display and print messages from other switch administrators and Avaya personnel, using the bulletin board. Anyone with the appropriate permissions can use the bulletin board for messages. Only one user can post or change a message at a time.

Whenever you log in, the system alerts you if you have any messages on the bulletin board and the date of the latest message. Also, if Avaya personnel post high-priority messages while you are logged in, you receive notification the next time you enter a command. This notification disappears after you enter another command and reoccurs at login until deleted by Avaya personnel.

You maintain the bulletin board by deleting messages you have already read. You cannot delete high-priority messages. If the bulletin board is at 80% or more capacity, a message appears at login indicating how much of its capacity is currently used (for example, 84%). If the bulletin board reaches maximum capacity, new messages overwrite the oldest messages.

NOTE:
The bulletin board does not lose information during a system reset at level 1 or level 2. If you save translations, the information can be restored if a system reset occurs at levels 3, 4, or 5.

Instructions

Displaying messages

To display the bulletin board:

1. Type **display bulletin-board** and press RETURN.

   The Bulletin Board screen appears.

```
Message (* indicates high-priority)                  Date
*Avaya is in the process of                         03/02/98
*investigating your trunk lockup problem.           03/02/98
*The Bulletin Board will be updated as              03/02/98
*we find information.                               03/02/98
*
*
*
*
*
*
We recently added a new trunk group (14)            03/02/98
and have had many of the members getting            03/02/98
locked up.                                          03/02/98
```

Posting a message

In our example, we post a message to the bulletin board about a problem with a new trunk group, and an Avaya representative replies to our message.

To post a message to the bulletin board:

1. Type **change bulletin-board** and press RETURN.

   The Bulletin Board screen appears. There are three pages of message space within the bulletin board. The first page has 19 lines, but you can only enter text on lines 11-19. The first 10 lines on page 1 are for high-priority messages from Avaya personnel and are noted with an asterisk (*). The second and third pages each have 20 lines, and you can enter text on any line. The system automatically enters the date the message was posted or last changed to the right of each message line.

2. Type your message.

   You can enter up to 40 characters of text per line. You also can enter one blank line. If you enter more than one blank line, the system consolidates them and displays only one. The system also deletes any blank line if it is line one of any page. You cannot indent text on the bulletin board. The TAB key moves the cursor to the next line.

3. Press ENTER to save your changes.

Deleting messages

To delete a message from the bulletin board:

1. Type **change bulletin-board** and press RETURN.

   The Bulletin Board screen appears.

2. Enter a space as the first character on each line of the message you want to delete and press RETURN.

3. Press ENTER to save your changes.

Saving translations

NOTE:
An alternate method of saving (backing up) and restoring translations is available for Avaya™ S8100 Media Server with CMC1 Media Gateway systems. See “Backup via the Web interface (Avaya™ S8100 Media Server only)” on page 53.

Avaya MultiVantage retains all translation data in memory while the system is operating. If the switch goes down, you lose all translation data. You must save in-memory translation data to the memory card (flash ROM), disk, or tape. Saving translation data to memory card or tape is the same as backing up your system.

NOTE:
Save translations on a daily basis. You may want to save translations after business hours to prevent dial tone delays or during slow business hours if your business is open 24 hours.

The save translation command writes two time-stamped identical copies of the translation data to the selected memory card, disk, or tape. The save writes one complete copy first, then writes the second copy in a different area of the device — both with the same time-stamp. Failure during a save, including a system crash, usually affects only one copy. The affected copy is marked “bad” and should not be used for backup.

You can set save translation to be completed automatically as part of regularly scheduled maintenance or manually, as needed. For more information about saving translations automatically, refer to the maintenance book for your system.

Tip:
To determine if your system saves translations automatically, type **display system-parameters maintenance** *to see if you have scheduled maintenance.*

Translation copy protection assigns a number to a specific phone system and to the flash card or set of flash cards that belong to that system. On an Avaya MultiVantage on a DEFINITY Server CSI or Avaya MultiVantage on a DEFINITY Server SI, this number is the same on both the translation storage device (flash card) and the Flash PROM (Programmable Read Only Memory) of the processor circuit pack. In a duplicated system, the Flash PROM of each processor circuit pack has a translation ID and both ID’s are stored on the memory card.

An attempt to initialize (boot) the system with translations that do not contain the same identification number as stored in the processor circuit pack raises a major alarm and disables access to the save translations command for all non-Avaya logins. You also receive a warning message on the copyright screen notifying you of the mismatch. Contact your Avaya representative to correct this mismatch and reset the save translations command. You must correct the mismatch before the end of the specified grace period, otherwise you cannot access system management commands (such as: add, change, remove, and duplicate) that modify translation data. Avaya specifies the grace period during a system installation or following an upgrade.

Before you start

If you are saving translations to a memory card or tape, you must verify the memory card or tape is in place and clear any active alarms from the alarms panel.

If your switch is an Avaya MultiVantage on a DEFINITY Server CSI or Avaya MultiVantage on a DEFINITY Server SI, verify the memory card translation ID matches the translation ID of your switch's Flash PROM.

Instructions

In our example, we save translations to the tapes on both processor A and B.

To save translations manually:

1. Type **save translation both tape** and press RETURN.

   The save process can take up to 10 minutes. You cannot administer your system while the save is in process. The Save Translation screen appears.

```
                              SAVE TRANSLATION


Processor                Command Completion Status          Error Code
SPE_A                             Success                        0
SPE_B                             Success                        0
```

2. If there is an error message in the Command Completion Status field and an error code in the Error Code field, clear the error and repeat the save process.

More information

When mass storage system (MSS) devices on both processors in a duplex system are specified, translation data is saved from the active processor to the active and standby MSS devices at the same time. If the save to one device fails or one device is out of service, the other save continues. You receive the status of each save separately.

NOTE:
If you have a duplex system and you save translation data to both MSS devices one at a time, translation data inconsistencies between the two devices can occur.

Fixing problems

NOTE:
You cannot see whether the translation ID on the flash card corresponds to the number on the Processor circuit packs. However, if the numbers do not match, the system issues an error message when you attempt a save translation operation.

When failures occur, the system responds with the following error codes.

Problem	Possible causes	Solution
1	Save translation cannot write to the active drive.	Repeat the save translation process for the active drive.
2	Save translation cannot write to the standby drive.	Repeat the save translation process for the standby drive.

For more information about error messages and how to correct them, refer to the maintenance book for your system.

Related topics

Refer to your maintenance book for information about backing up or restoring your system.

Refer to ‘‘Saving announcements’’ on page 437 for information about backing up announcements for your system.

Refer to ‘‘Restoring announcements’’ on page 439 for information about restoring announcements to your system.

Backup via the Web interface (Avaya™ S8100 Media Server only)

The following are web interface procedures:

1. Open internet explorer.
1. Enter **http://<IP address>** in the address area of the web browser.

   The DEFINITY ONE Home page displays:
2. Click **Administer System**.

   The login screen displays:

   Enter your login ID and password.

The login ID must have the correct backup permissions and be a member of the Avaya™ S8100 Media Server Administrator's login group.

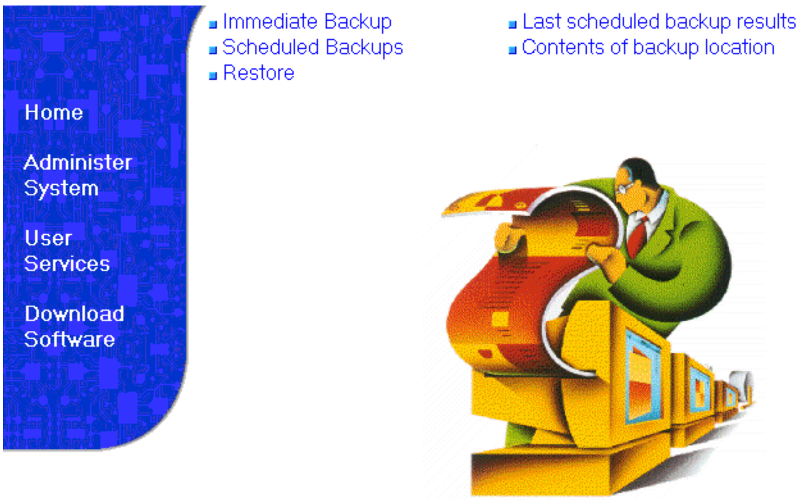
The Notice screen displays:

3. Click **Continue**.

   The System Administration screen displays:

   Click **Backup and Restore** to open main backup menu.

Backup and restore main menu (Avaya™ S8100 Media Server only)



From the backup and restore main menu, you can:

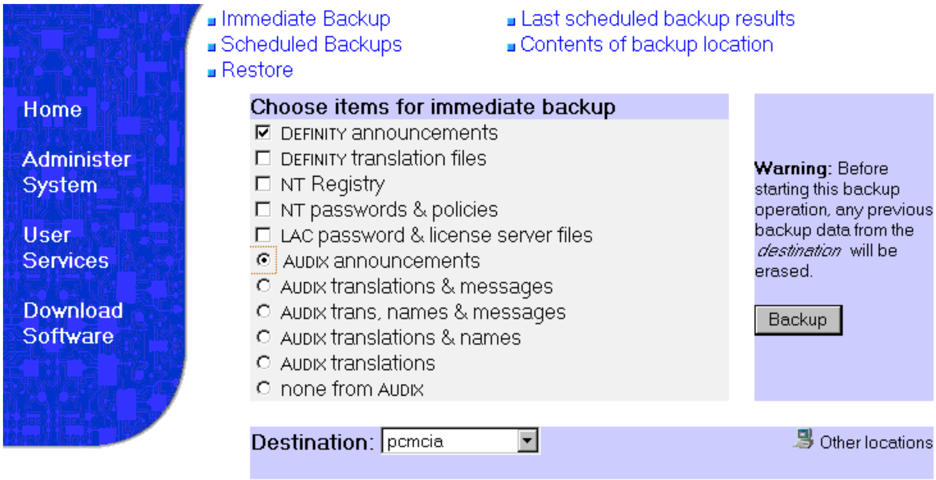
- Perform immediate backups
- Schedule multiple backups
- Restore backups
- Access last scheduled backup information
- View contents of backup location

NOTE:
As you navigate the backup and restore screens, the main menu items remain available. If you are using the Web, use the Back button to return to previous screens.

Perform immediate backup

To perform an immediate backup, click **Immediate Backup**.

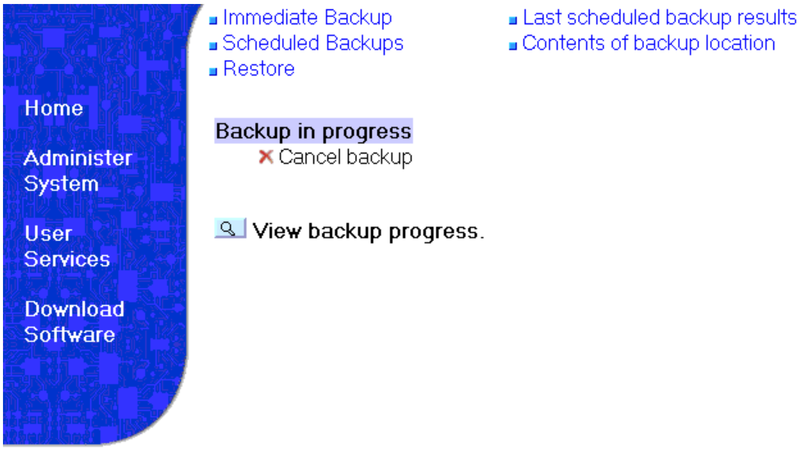
The following screen displays:



1. From the Destination menu, select a backup destination.This can be a LAN address or a PCMCIA Flash Disk
2. Select items for immediate backup.
3. .Click **Backup**.

NOTE:
When backing up to a LAN address, a shared drive must be installed on a non- Avaya™ S8100 Media Server.

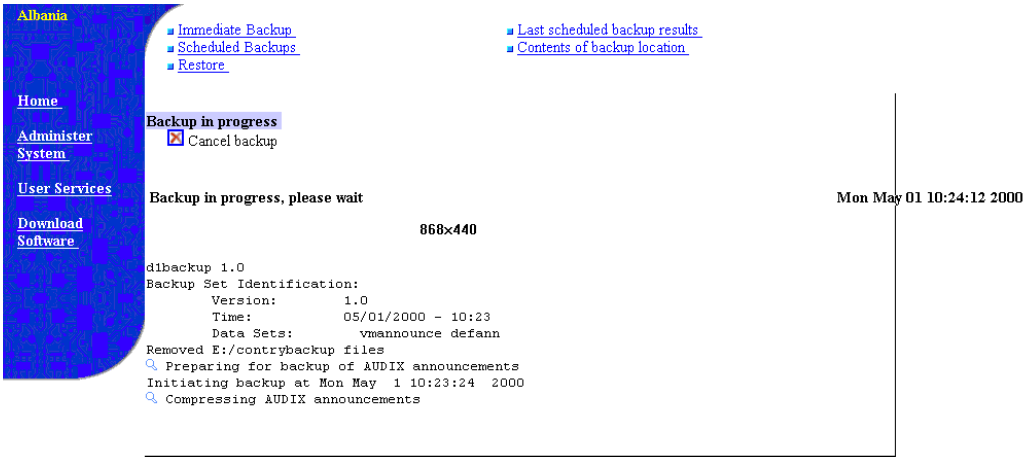
After you click **Backup**, the following screen displays:



Viewing backup progress

To view backup progress, click **View Backup Progress**.

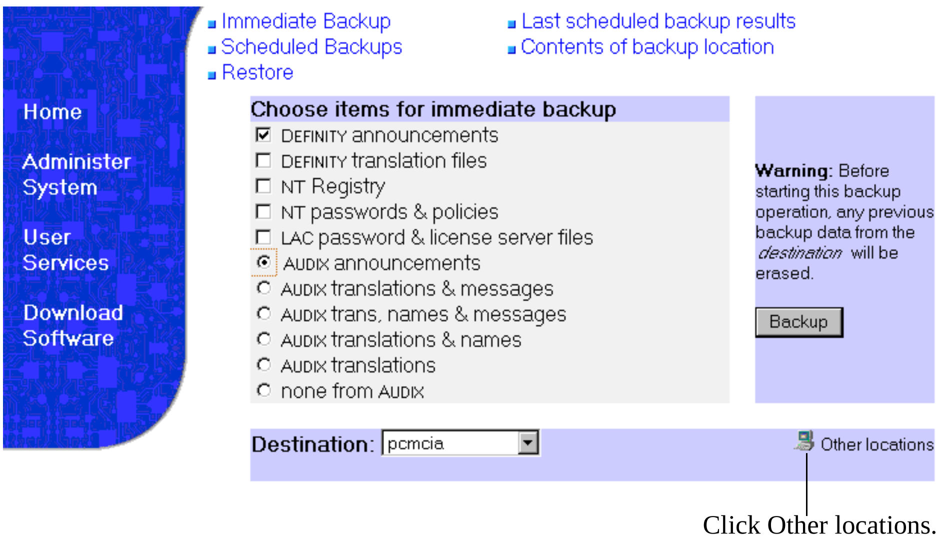
The following screen displays:



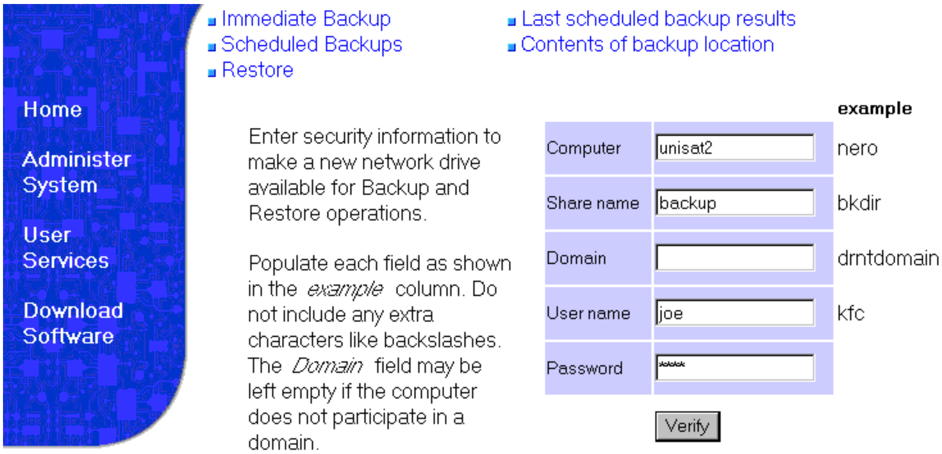
Backing up to a LAN address

You can back up your data to a LAN address using the Other locations feature. To back up data to a LAN address:

1. Click **Other locations**.

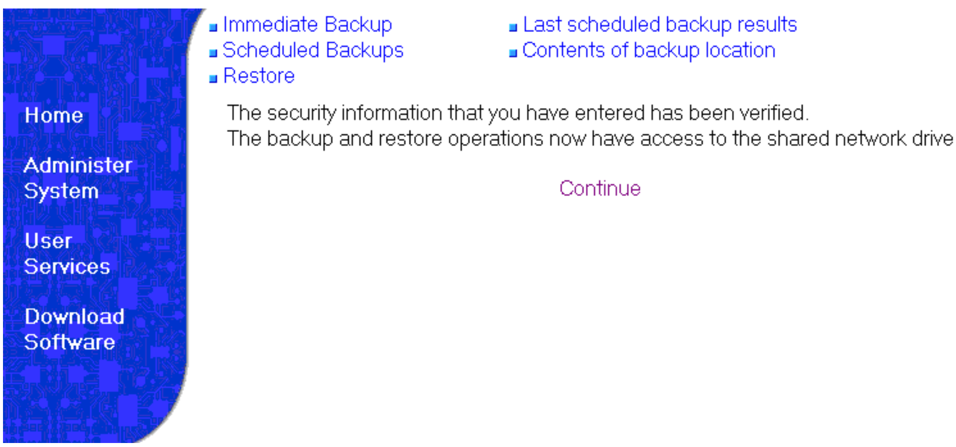


The following screen displays:



2. Enter LAN location information.
3. Click **Verify**.

   The following screen displays:



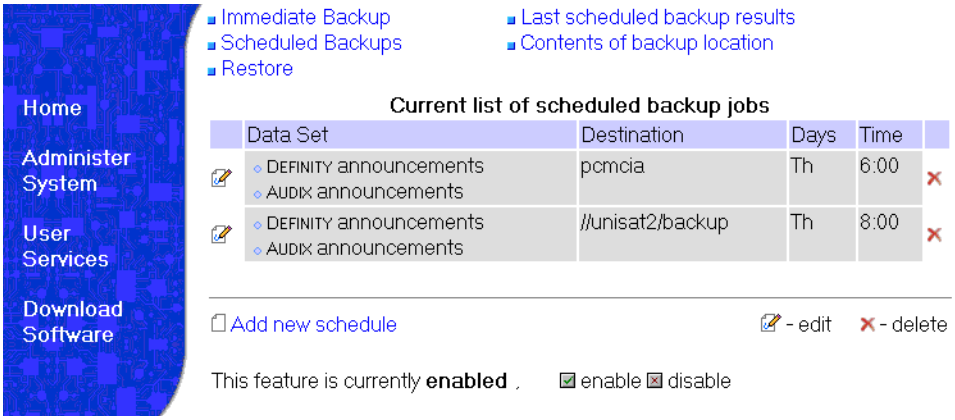
4. Click **Continue** to return to the Immediate backup screen.
5. Select items to back up and select **Backup**.

Viewing scheduled backups

To view scheduled backups:

1. Click **Scheduled Backups**.

   The following screen displays:



   From this screen, you can add, edit, or delete scheduled backups.

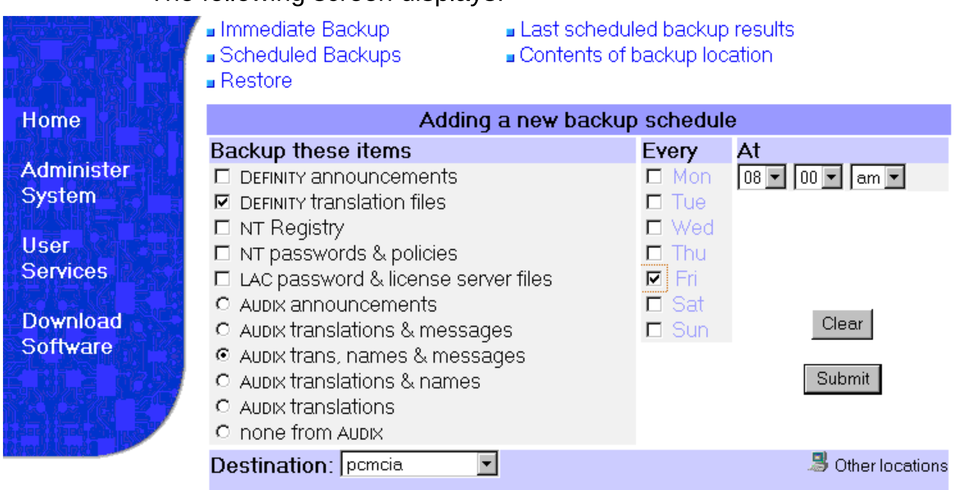
NOTE:
The backup feature can be disabled and later enabled to allow you to perform another function. If disabled, the current schedules remain intact.

Adding a new scheduled backup (multiple backup schedules)

To add a new scheduled backup to the list:

1. On the Current list of scheduled backup jobs screen, click **Add new schedule**.

   The following screen displays:

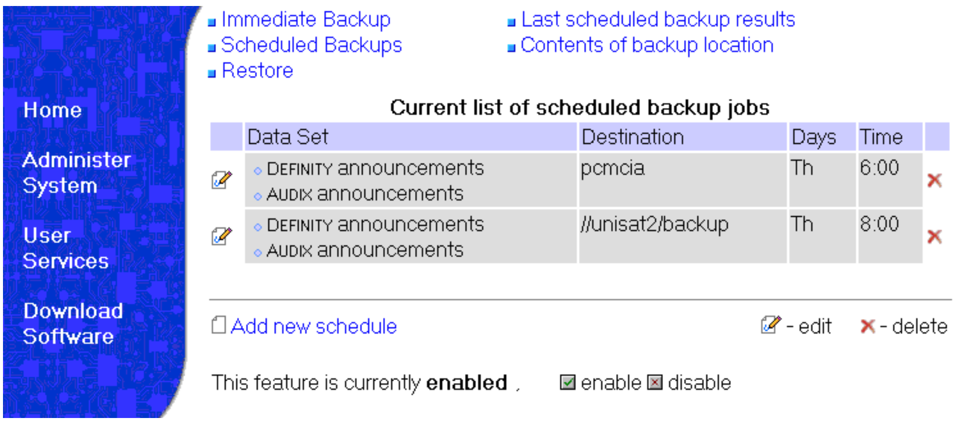


2. Select backup destination either to a LAN address or a PCMCIA Flash Disk.
3. Select items for scheduled backup.
4. Select a day and time for the backup.
5. Click **Submit**.

Changing a scheduled backup

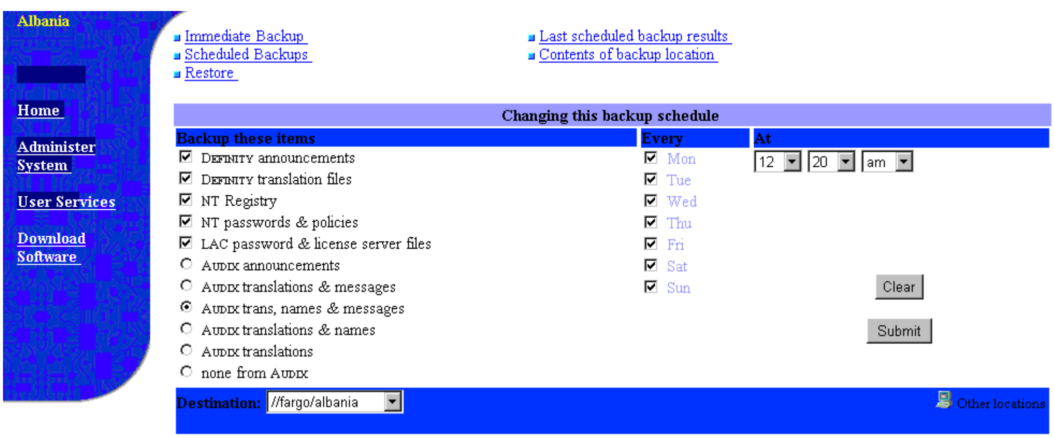
1. To change an existing scheduled backup, click on Scheduled Backups.

The Current List of scheduled backup jobs displays:



2. Click on the pencil symbol next to the scheduled backup you want to change.

   The Changing this backup schedule displays:



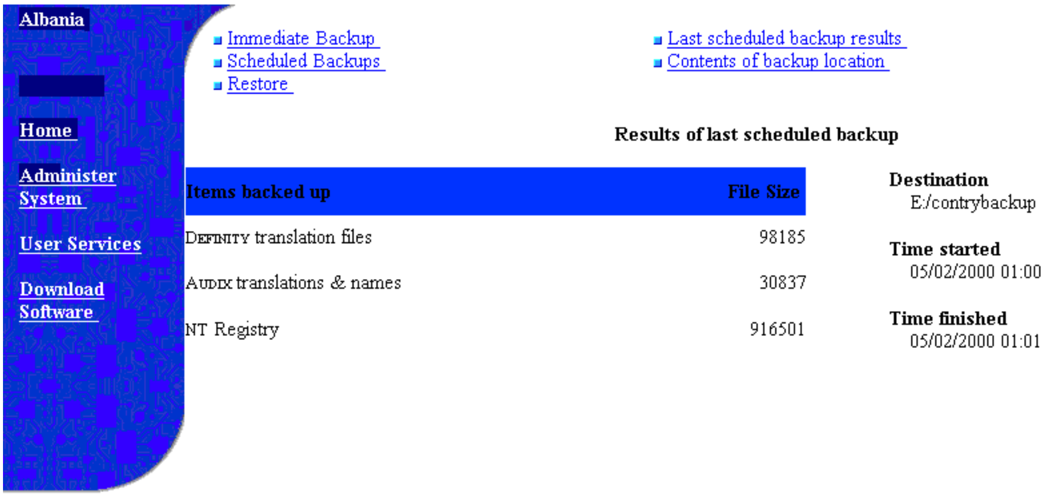
3. Make any changes as appropriate, and click Submit.

   The Current list of scheduled backups displays. The changes are activated.

Checking Scheduled Backup Status

To review previous backups, click **Last scheduled backup results**.

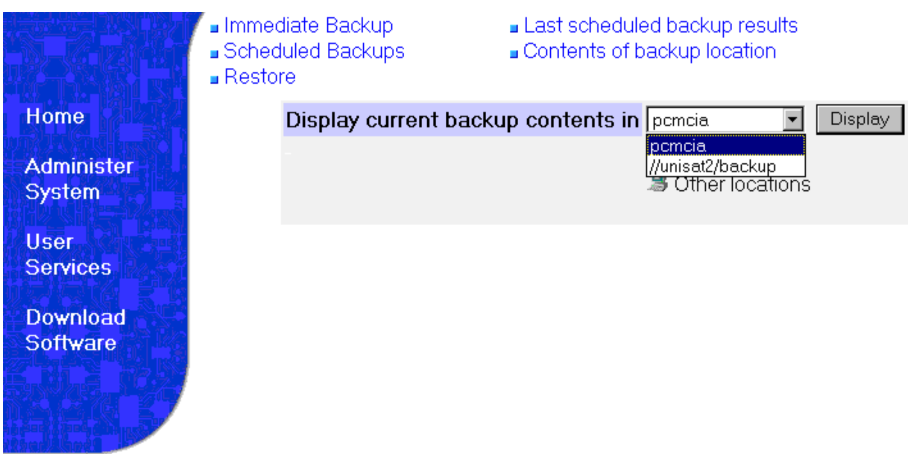
The following screen displays:



Checking the contents of a backup

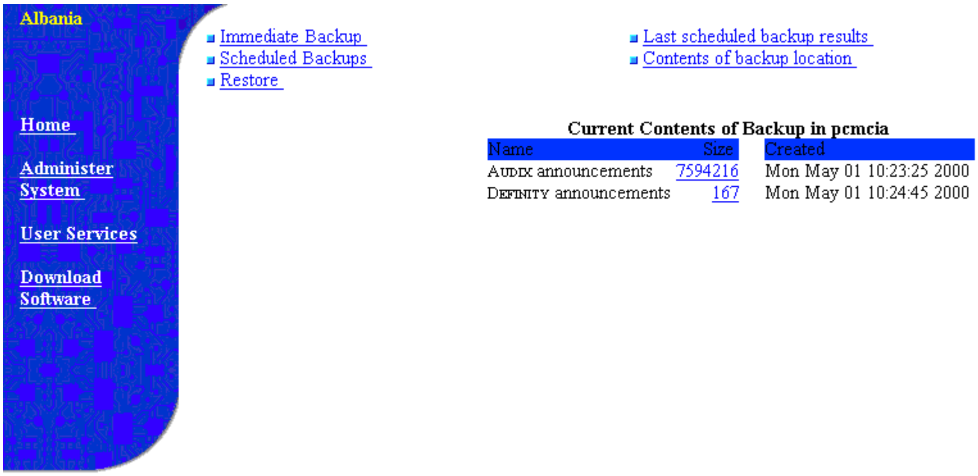
To view contents click **Contents of backup location**.

The following screen displays:



Scroll to the location of backup contents and click **Display** or click **Other locations.**

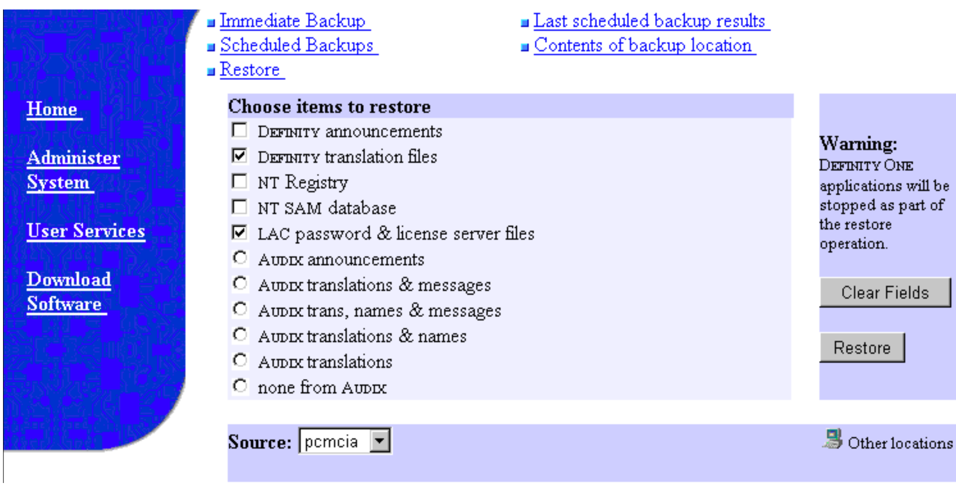
The following screen displays:



Perform restore

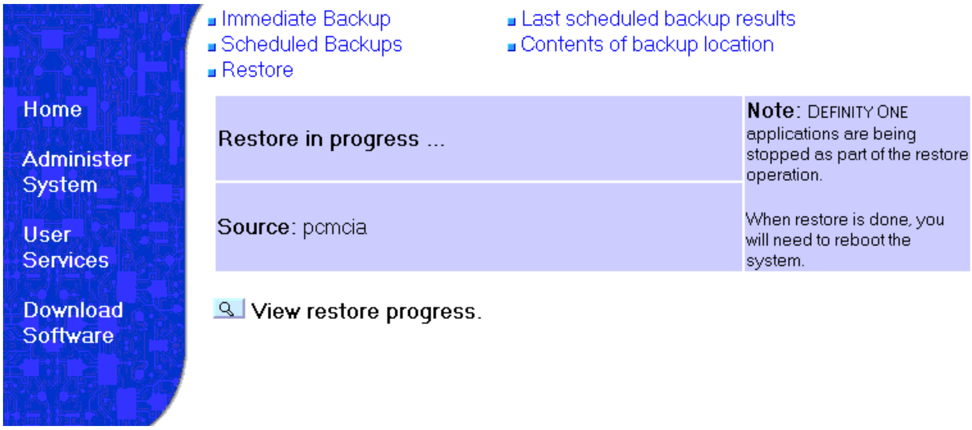
1. Click **Restore**.

   The following screen displays:



2. Select the restore source from the Source pull down menu. The source is the media and/or location of the backup data. It could be a network or shared drive location, or the pcmcia flash card.
3. Select items to restore.
4. Click **Restore**.

The following screen displays:



5. After the restore is completed, reboot the system.

NOTE:
A reboot is required. The restored stations will not be used if the Avaya™ S8100 Media Server is not rebooted.

Introduction to the MultiVantage system

2

This section provides you with general information about Avaya MultiVantage and some of the system-wide functions. It explains how to understand your configuration, read and use your dial plan, and shows you how to make simple changes such as adding extension ranges. This section also explains how to assign feature access codes (FAC).

Understanding your configuration

At a very basic level, the Avaya MultiVantage consists of hardware to perform call processing, and the software to make it run. You use the administration interface to let the system know what hardware you have, where it is located, and what you want the software to do with it.

You can find out which circuit packs are in the system and which ports are available by entering the command **list configuration all**. There are variations on this command that display different types of configuration information. Use the help function to experiment, and see which command works for you.

1. To view a list of port boards on your system, type **list configuration port-network** and press RETURN.

   The System Configuration screen appears.

```
                              SYSTEM CONFIGURATION
Board                                                    Assigned Ports
Number     Board Type Code     Vintage     u=unassigned t=tti p=psa

01A05      DIGITAL LINE          TN754B 000002      01 u  03 u  05 u  07 08
01A06      ANALOG LINE           TN742  000010      01 02 03 04 u  u  u  u
01B05      ANALOG LINE           TN746B 000008      u  u  u  u  u  u  u  u
                                                    u  u  u  u  u  u  u  u
01C04      ANALOG LINE           TN746B 000008      u  u  u  u  u  u  u  u
                                                    u  u  u  u  u  u  u  u
01C05      DIGITAL LINE          TN2224 000004      01 u  u  04 u  u  07 08
                                                    u  u  u  u  u  u  u  u
                                                    u  u  u  u  u  u  u  u
01C06      HYBRID LINE           TN762B 000004      01 02 u  u  u  u  u  u
01C10      DIGITAL LINE          TN754  000004      u  u  u  u  u  u  u  u
```

The System Configuration screen shows all the boards on your system that are available for connecting phones, trunks, data modules and other equipment. You can see the board number, board type, circuit-pack type, and status of each board's ports. The **u** entries on this screen indicate unused ports that are available for you to administer. These may also appear as **p** or **t**, depending on settings in your system.

You will find many places in the administration interface where you are asked to enter a port or slot. The port or slot is actually an address that describes the physical location of the equipment you are using.

A port address is made up of four parts:

- cabinet — the main housing for all the switch equipment. Cabinets are numbered starting with 01.
- carrier — the rack within the cabinet that holds a row of circuit packs. Each carrier within a cabinet has a letter, A–E.
- slot — the space in the carrier that holds an individual circuit pack. Slots are numbered 01-16.
- port — the wire that is connected to an individual piece of equipment (such as a phone or data module). The number of ports on a circuit pack varies depending on the type.

So, if you have a single-carrier cabinet, the circuit pack in slot 06 would have the address 01A06. If you want to attach a phone to the 3rd port on this board, the port address is 01A0603 (01=cabinet, A=carrier, 06=slot, 03=port).

Understanding the dial plan

Your dial plan tells your system how to interpret dialed digits. For example, if you dial 9 on your system to access an outside line, it is actually the dial plan that tells the system to find an external trunk when a dialed string begins with a 9.

The dial plan also tells the system how many digits to expect for certain calls. For example, the dial plan may indicate that all internal extensions are 4-digit numbers that start with 1 or 2.

Let's take a look at an example dial plan so you'll know how to read your system's dial plan. The following figure shows an example of a simple dial plan.

```
change dialplan analysis                                         Page   1 of 1
                             DIAL PLAN ANALYSIS TABLE
                                                                    Percent Full:    9


   Dialed  Total  Call                                Dialed  Total  Call
   String  Length Type                                String  Length Type
     0       1    attd                                  __      _    ____
     1       3    dac                                   __      _    ____
     3       5    ext                                   __      _    ____
     4       6    ext                                   __      _    ____
     5       7    ext                                   __      _    ____
     *       3    fac                                   __      _    ____
     #       3    fac                                   __      _    ____
```

The Dial Plan Analysis Table defines the dialing plan for your system.

The Call Type column in the Dial Plan Analysis Table indicates what the system does when a user dials the digit or digits indicated in the Dialed String column. The Total Length column indicates how long the dialed string will be for each type of call. For example, this dial plan shows that when users dial a 5-digit number that starts with 3, they are dialing an extension.

The Dial Plan Analysis Table in our example contains the following call types:

- Attendant (attd) — Defines how users call an attendant. Attendant access numbers can be any number from 0 to 9 and contain 1 or 2 digits.

  In our example figure, the system calls an attendant when users dial 0.

- Dial access code (dac) — Allows you to use trunk access codes (TAC) and feature access codes (FAC) in the same range. For example, you could define the group 100–199, which would allow both FAC and TAC in that range. Dial access codes can start with any number from 1 to 9, * and #, and contain up to 4 digits.

  In our example figure, dial access codes begin with 1 and must be 3 digits long.

  NOTE:
  The Dial Plan Analysis Table does not allow you to enter a range specifically for trunk access codes. However, the Trunk Group screen still allows you to assign a TAC to a trunk group. The TAC you enter on the Trunk Group screen must match the format you have administered for a DAC on the Dial Plan Analysis Table.

- Extensions (ext) — Defines extension ranges that can be used on your system. In our figure, extensions must be in the ranges 30000–39999, 40000–49999 and 50000–59999.
- Feature access codes (fac) only — FAC can be any number from 1 to 9 and contain up to 4 digits. You can use * or #, but only as a first digit. In our example, feature access codes can begin with * or # and are 3-digits long.

The Dial Plan Analysis Table works with the Dial Plan Parameters Table for fully defining your dial plan. The Dial Plan Parameters Table allows you to set system-wide parameters for your dial plan.

Displaying your dial plan

You might want to take this opportunity to look at and interpret your own dial plan. To display your system's dial plan:

1. Type **display dialplan analysis** and press RETURN.

Modifying your dial plan

It is easy to make changes to your dial plan. For example, let's add a new range of dial access codes to the dial plan. We want to be able to assign both FAC and TAC in the 700–799 range.

1. Type **change dialplan analysis** and press RETURN.

   The Dial Plan Analysis appears.

2. Move the cursor to an empty row.
3. Type **7** in the Dialed String column. Press TAB to move to the next field.
4. Type **3** in the Total Length Column. Press TAB to move to the next field.

5. Type **dac** in the Call Type Column.
6. Press ENTER to save your changes.

Adding extension ranges

You may find that as your needs grow you want a new set of extensions. Before you can assign a station to an extension, the extension must belong to a range that is defined in the dial plan. Let's add a new set of extensions that start with 3 and are 4 digits long (3000–3999).

To add this set of extensions to the dial plan:

1. Type **change dialplan analysis** and press RETURN.

   The ''Dial Plan Analysis Table'' on page 717 appears.
2. Move the cursor to an empty row.
3. Type **3** in the Dialed String column. Press TAB to move to the next field.
4. Type **4** in the Total Length Column. Press TAB to move to the next field.
5. Type **ext** in the Call Type Column.
6. Press ENTER to save your changes.

Uniform dial plan

You can also set up a Uniform Dialing Plan that can be shared among a group of switches. For more information, see ''Uniform Dial Plan'' on page 1792.

Other options for the dial plan

You can establish a dial plan so that users only need to dial one digit to reach another extension. You can also establish a dial plan that allows users to dial, for example, three digits to reach one extension, and four digits to reach another. This is particularly useful in the hospitality industry, where you want users to be able to simply dial a room number to reach another guest. For more information, see ''Single-Digit Dialing and Mixed Station Numbering'' on page 1701.

Adding feature access codes

As your needs change, you may want to add a new set of feature access codes (FAC) for your system. Before you can assign a FAC on the Feature Access Code screen, it must conform to your dial plan.

In our example, if you want to assign a feature access code of 33 to Last Number Dialed, first you need to add a new FAC range to the dial plan.

To add a FAC range from 30–39:

1. Type **change dialplan analysis** and press RETURN.

   The ''Dial Plan Analysis Table'' on page 717 appears.
2. Move the cursor to an empty row.
3. Type **3** in the Dialed String column. Press TAB to move to the next field.
4. Type **2** in the Total Length Column. Press TAB to move to the next field.
5. Type **fac** in the Call Type Column.
6. Press ENTER to save your changes.

Changing feature access codes

Feature access codes (FAC) allow users to activate and deactivate features from their phones. A user who knows the FAC for a feature does not need a programmed button to use the feature. For example, if you tell your users that the FAC for the Last Number Dialed is *33, then users can redial a phone number by entering the FAC, rather than requiring a Last Number Dialed button.

Many features already have factory-set feature access codes. You can use these default codes or you can change them to codes that make more sense to you. However, every FAC must conform to your dial plan and must be unique. For more information about the dial plan, refer to ''Understanding the dial plan'' on page 65.

Let's try an example. If you want to change the feature access code for Call Park to *72:

1. Type **change feature-access-codes** and press RETURN.

   The Feature Access Code screen appears.
2. Move the cursor to the Call Park Access Code field.
3. Type ***72** in the access code field over the old code.
4. Press ENTER to save your changes.

   If you try to enter a code that is assigned to a feature, the system warns you of the duplicate code and does not allow you to proceed until you change one of them.

Tip:
To remove any feature access code, merely delete the existing FAC and leave the field blank.

Controlling the calls your users can make and receive

The Avaya MultiVantage provides several ways for you to restrict the types of calls your users can make, and the features that they can access.

You use Class of Restriction (COR) to define the types of calls your users can place and receive. Your system may have only a single COR, a COR with no restrictions, or as many CORs as necessary to effect the desired restrictions.

You will see the COR field in many different places throughout the MultiVantage system - when administering phones, trunks, agent logins, and data modules, to name a few. You must enter a COR on these screens, although you control the level of restriction the COR provides.

Strategies for assigning CORs

The best strategy is to make it as simple as possible for you and your staff to know which COR to assign when administering your system. You can create a unique COR for each type of user or facility, for example, call center agents, account executives, administrative assistants, WATS trunks, paging zones or data modules.

You can also create a unique COR for each type of restriction, for example, toll restriction, or outward restriction. If you have a number of people who help you administer your system, using this method would also require the additional step of explaining where you wanted to use each type of restriction. See ''CLAN and processor CLAN QoS and CIDR support'' on page 1521 for more information.

To find out what CORs are administered in your system already, type **list cor.** You can also display information for a single COR by typing **list cor #**.

Allowing users to change CORs

You can allow specific users to change their class of restriction COR from their telephones using a Change COR feature access code. You can also limit this feature by insisting that the user enter a password as well as a feature access code before they can change their COR.

Before you start

- Be sure that Change COR by FAC field is set to y on the System-Parameters Customer-Options screen. Note that you cannot have both Change COR by FAC and Tenant Partitioning enabled.

- Be sure that each user (who you want to allow to change a COR) has a class of service with console permissions. For more information about console permissions, refer to ''Class of Service'' on page 650.

Instructions

To allow users to change their own class of restriction, you must define a feature access code and can, optionally, create a password. For example, let's create a change COR feature access code of *55 and a password of 12344321.

1. Type **change feature-access-codes** and press ENTER.

   The Feature Access Code screen appears.

2. Move the cursor to the Change COR Access Code field.
3. Type ***55** in the access code field.
4. Press ENTER to save your changes.

   Now we have to define the password.

5. Type **change system-parameters features** and press ENTER.

   The Feature-Related System Parameters screen appears.

```
change system-parameters features                                   Page 10 of 10
                         FEATURE-RELATED SYSTEM PARAMETERS

AUTOMATIC EXCLUSION PARAMETERS

                       Automatic Exclusion by COS? y
                Automatic Exclusion Coverage/Hold? y
            Automatic Exclusion with Whisper Page? y

                    Password to Change COR by FAC: 12344321
```

6. Move to the Password to Change COR by FAC field and enter **12344321**.

   This field determines whether or not the MultiVantage system requires the user to enter a password when they try to change their COR. Avaya recommends that you require a password.

7. Press ENTER to save your changes.

Controlling the features your users can access

The Avaya MultiVantage offers a wide range of features and functions. Some of these you can administer differently from one user to the next. For example, you can give one user a certain set of phone buttons, and the next user a completely different set, depending on what each person needs to get his/her job done. You decide on these things as you administer the phones for these individuals. Refer to ‘‘Managing phones’’ on page 85 for more information.

You can also establish classes of service (COS) to control the switch features that users can access. For example, you can permit users to forwarding their calls, or restrict them from placing priority calls. Once you have set permissions for a class of service, you assign this COS to a user’s telephone or other device.

Classes of service are very similar to classes of restriction. COR and COS do not overlap in the access or restrictions they control. Refer to ‘‘Class of Service’’ on page 650 for more information.

Class of service and class of restriction give you great flexibility with what you allow users to do. If you are in doubt about the potential security risks associated with a particular permission, read further in this document, consult the *Avaya Products Security Handbook*, or contact your Avaya representative.

System-wide settings

There are some settings that you enable or disable for the entire system, and these settings effect every user. You may want to look over the various System Parameters screens and decide which settings best meet the needs of your users.

To see a list of the different types of parameters that control your system, type **display system-parameters** and press HELP. You can change some of these parameters yourself. Type **change system-parameters** and press HELP to see which types of parameters you can change. In some cases, an Avaya representative is the only person who can make changes, such as to the System-Parameters Customer-Options screen.

This chapter offers a few examples of how you establish these system-wide settings. The Screen reference contains explanations of each of the system parameters screens, and how to use them.

Changing feature parameters

You can modify the system parameters that are associated with some of the system features. For example, you can use the system parameters to allow music to play if callers are on hold or to allow trunk-to-trunk transfers on the system.

Generally, Avaya sets your system parameters when your system is installed. However, you can change these parameters as your organization's needs change. For example, let's say that you are told that the number of rings between each point for new coverage paths should change from 4 to 2 rings.

To change the number of rings:

1. Type **change system-parameters coverage/forwarding** and press RETURN.

   The System Parameters Call Coverage / Call Forwarding screen appears.

```
                                                              Page 1 of 2
                SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

CALL COVERAGE/FORWARDING PARAMETERS
           Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 2
         Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 2
                           Coverage - Caller Response Interval (seconds): 4


COVERAGE
                                         Keep Held SBA At Coverage Point? y
              External Coverage Treatment For Transferred Incoming Calls? n


FORWARDING
                                                   Call Forward Override? n
                                               Coverage After Forwarding? y
```

2. In the Local Coverage Subsequent Redirection/CFWD No Answer Interval field, type **2.**
3. Press ENTER to save your changes.

   Each phone in a Call Coverage path now rings twice before the call routes to the next coverage point. The Local Cvg Subsequent Redirection/CFWD No Ans Interval field also controls the number of rings before the call is forwarded when you use Call Forwarding for busy/don't answer calls. This applies only to calls covered or forwarded to local extensions. Use Off-Net to set the number of rings for calls forwarded to public network extensions.

Administering treatment for denied or invalid calls

You can administer your system to reroute denied or invalid calls to an announcement, the attendant, or to another extension.

Instructions

In this example, we want:

- all outward restricted call attempts to route to an announcement at extension 2040
- all incoming calls that are denied to route to the attendant
- all invalid dialed numbers to route to an announcement at extension 2045

1. Type **change system-parameters features** and press RETURN.

   The Feature-Related System Parameters screen appears.
2. In the Controlled Outward Restriction Intercept Treatment field, type **announcement**.
3. Another blank field appears.
4. In this blank field, type **2040**.
5. This is the extension of an announcement you recorded earlier.
6. In the DID/Tie/ISDN Intercept Treatment field, type **attd**.
7. This allows the attendant to handle incoming calls that have been denied.
8. In the Invalid Number Dialed Intercept field, type **announcement**.
9. Another blank field appears.
10. In this blank field, type **2045**.
11. This is the extension of an announcement you recorded earlier.
12. Press ENTER to save your changes.

Setting up Music-on-Hold

Music-on-Hold automatically provides music to a caller placed on hold. Providing music lets the caller know that the connection is still active. The system does not provide music to callers in a multiple-party connection who are in queue, on hold, or parked.

Before you start

You need to determine the music source you will use, and obtain the necessary circuit pack. Refer to AVAYA MultiVantage™ Hardware Solutions Guide for more information about required hardware.

NOTE:
If you use equipment that rebroadcasts music or other copyrighted materials, you may be required to obtain a copyright license from or pay fees to a third party such as the American Society of Composers, Artists, and Producers (ASCAP) or Broadcast Music Incorporated (BMI).

Instructions

In this example, we administer the system to allow local callers and incoming trunk callers to hear music while on hold. Note that if you use Tenant Partitioning, you cannot set up Music on Hold this way. See ''Providing service for multiple tenants'' on page 75 for more information.

1. Type **change system-parameters features** and press RETURN.

   The Feature-Related System Parameters screen appears.

```
change system-parameters features                               Page 1 of 9
                         FEATURE-RELATED SYSTEM PARAMETERS


                                 Trunk-to-Trunk Transfer? none
 Automatic Callback - No Answer Timeout Interval (rings): 4_
                  Call Park Timeout Interval (minutes): 10
     Off-Premises Tone Detect Timeout Interval (seconds): 20_
                              AAR/ARS Dial Tone Required? y
                                      Music/Tone On Hold: music  Port: _______
           Music (or Silence) On Transferred Trunk Calls: all
                        DID/Tie/ISDN Intercept Treatment: attd
               Messaging Service Adjunct (MSA) Connected? y
 Internal Auto-Answer of Attd-Extended/Transferred Calls: y
               Automatic Circuit Assurance (ACA) Enabled? y
                                      ACA Referral Calls: local
                                ACA Referral Destination: ________
            ACA Short Holding Time Originating Extension: ______
             ACA Long Holding Time Originating Extension: ______
          Abbreviated Dial Programming by Assigned Lists:
     Auto Abbreviated/Delayed Transition Interval(rings):
```

2. In the Music/Tone On Hold field, type **music**.

   The Port field appears.

3. In the Port field, type **6040**.

   This is the port address of the music source.

4. In the Music (or Silence) on Transferred Trunk Calls, type **all**.

5. Press ENTER to save your changes.

6. Now administer a class of restriction with Hear System Music on Hold set to **y**, to allow your local users to hear music on hold.

More information

If a call with either Data Privacy or Data Restriction activated is placed on hold, the Music/Tone on Hold is withheld. This is to prevent transmission of a musical tone that a connected data service might falsely interpret as a data transmission.

If you administer the Music/Tone on Hold field to provide music, the system provides the music after a hunt group or ACD split delayed announcement.

Music on Hold may sound distorted when played to IP trunks or to IP phones through certain codecs, particularly the G.723 codec. You can provide different on-hold materials for these endpoints. Using the instructions for ‘‘Providing service for multiple tenants’’, create one tenant partition for all endpoints that do not use the G.723 codec and administer Music on Hold for this tenant. Create another tenant partition for endpoints that use the G.723 codec and administer silence, news, or other material that does not sound distorted for these endpoints.

Providing service for multiple tenants

If you manage the switching system for an entire office building, you may need to provide individualized phone service for each of the firms who are tenants. You can set up your system so that each tenant can have its own attendant, and can chose to have music or play special announcements while callers are on hold.

Before you start

Before you can administer tenants in your system, Tenant Partitioning must be enabled on the System-Parameters Customer-Options screen. To allow tenants to select their own Music on Hold, you must purchase and administer equipment. Refer to AVAYA MultiVantage™ Hardware Solutions Guide for more information about required hardware. Your Avaya representative can help you administer this hardware.

Instructions

In this example, we are going to administer the system to allow one tenant to play Country music for callers on hold, and another to play Classical music. We will assign these music types to two new tenants.

1. Type **change music-sources** and press return.

   The Music Sources screen appears.

```
                              Music Sources                        Page 1 of X

           Source    Type      Port       Description

             1      music     01A1001    Country
             2      tone                 Tone-on-Hold
             3      music     01A1003    Classical
             4      none
             5      none
             6      none
             7      music     12B1301    Oldies
             8      none
             9      none
            10      none
            11      music     04C2003    Rock
            12      none
            13      none
            14      none
            15      none
```

2. For Source 1, enter **music** for the Type, **01A1001** for the Port, and **Country** for the Description.
3. Move to Source 3, and enter **music** for the Type, **01A1003** for the Port, and **Classical** for the Description.
4. Press ENTER to save your changes.

5. Type **change tenant 1** and press RETURN.

   The Tenant screen appears.

```
                        Tenant 18

     Tenant Description: ________________________________________

        Attendant Group: 1

  Ext Alert Port (TAAS): _______ Ext Alert (TAAS) Extension: ____

      Night Destination: _____

           Music Source: 1
```

6. In the Tenant Description field, type **Dentist**.

   This identifies the client in this partition.

7. In the Attendant Group field, type **1**.

   The attendant group number must also appear in the Group field of the attendant console screen for this tenant.

8. In the Music Source field, type **1**.

   Callers to this tenant will now hear country music while on hold.

9. Press ENTER to save your changes.

10. To administer the next partition, type **change tenant 2** and press RETURN.

11. Administer this tenant, Insurance Agent, to use Attendant Group 2 and Music Source 3. Be sure to change the Attendant Console screen so that this attendant is in group 2.

12. This tenant's callers will hear classical music on hold.

Receiving notification in an emergency

If one of your users calls an emergency service such as the police or ambulance, someone, perhaps the receptionist, security or the front desk, needs to know who made the call. Thus, when the emergency personnel arrive, they can be directed to the right place.

You can set up the switch to alert the attendant and up to ten other extensions whenever an end-user dials an emergency number. The display on the notified user's phone shows the name and number of the person who placed the emergency call. The phones also ring with a siren-type alarm, which users must acknowledge to cancel.

Before you start

Decide if you want one user to be able to acknowledge an alert, or if all users must respond before an alert is cancelled.

Verify that ARS is enabled on the System-Parameters Customer-Options screen.

Make sure that the extensions you notify belong to physical digital display phones. Refer to *"Phone reference'' on page 1273* for a list of phone types. When you assign crisis alert buttons to the phones, check the type field on the station screen to be sure you are not using a virtual extension.

Instructions

In this example, we will set up the system to notify the attendant and the security guards at all 3 entrances when someone dials the emergency number 5555. All three guards must acknowledge the alert before it is silent.

1. Type **change ars analysis 5** and press RETURN.

   The AAR and ARS Digit Analysis Table appears.

```
                ARS DIGIT ANALYSIS TABLE
                        Location: all          Percent Full:   6

     Dialed          Total    Route     Call    Node  ANI
   String            Min Max  Pattern    Type    Num    Reqd
5555_____________     4_  4_     1____     alrt    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
_________________     __  __    _____     ____    ___     n
```

2. In the dialed string field, type **5555**.

   This is the number that end-users dial to reach emergency services.

3. In the Total Min and Max fields, type **4**.

   In this example, the user must dial all 4 digits for the call to be treated as an emergency call.

4. In the Route Pattern field, type **1**.

   In this example, we use route pattern 1 for local calls.

5. In the Call Type field, type **alrt**.

   This identifies the dialed string 5555 as one that activates emergency notification.

6. Press ENTER to save your changes.

Now set up the attendant console to receive emergency notification.

7. Type **change attendant 1** and press RETURN.

   The Attendant Console screen appears.

8. In the feature button area, assign a **crss-alert** button.

9. Press ENTER to save your changes.

10. Assign a **crss-alert** button to each security guard's phone.

    You cannot assign this button to a soft key. See ''Adding feature buttons'' on page 127 for more information.

Finally, we make sure that all security personnel and the attendant will have to acknowledge the alert.

11. Type **change system-parameters crisis-alert** and press RETURN.

    The “Crisis Alert System Parameters” screen appears.

12. Go to the Every User Responds field and type **y**.

13. Press ENTER to save your changes.

More information

Attendants cancel an alert by pressing the crisis alert button three times. The first button push turns off the siren, the second stops the lamp from flashing, and the third clears the display.

Digital phone users cancel the siren by pushing the crisis alert button. If you have set the system so that only one user needs to respond, this stops the alerting at all phones. If all users must respond, each phone continues to alert until that user presses the crisis alert button to acknowledge the alarm. The emergency caller’s name and extension remain on the display at this point. To completely cancel an alert and clear their displays, users press the Normal button.

Once you administer Crisis Alert, the switch still records each emergency call and sends a record to the journal printer, if available. If not, you can view the emergency log with the command **list emergency**.

Related topics

Refer to “Crisis Alert” on page 1536 for more detailed information.

For information on other ways to reach the attendant in an emergency, refer to “Emergency Access to the Attendant” on page 1548.

To determine what types of digital phones have displays, refer to “Phone reference” on page 1273.

To ensure that you can still make necessary phone calls in an emergency, refer to “Emergency Calls” on page 1551.

For information on setting up Centralized Automatic Message Accounting (CAMA) trunks and provide Caller’s Emergency Service Identification (CESID) information to your local community’s Enhanced 911 system, see “CAMA Numbering Format”.

For information about a way to work around updating the USA Automatic Location Identification data base, see Emergency Location extension field in ‘‘Station’’ on page 1066.

For information on how to administer IP phones to make emergency calls, see ‘‘Setting up emergency calls on IP phones’’ on page 107.

Notifying a digital pager of an emergency

You also have the option of having your emergency calls go to a digital pager. When someone dials an emergency number (for example, 911), the system sends the extension and location (that originated the emergency call) to the administered pager.

Before you start

- You need to administer a CRSS-ALRT button on at least one of the following.
  - — Attendant Console (use the **change attendant** command)
  - — Digital telephone set (use the **change station** command)
- The ARS Digit Analysis Table must have emergency numbers in the Call Type column set to **alrt** (crisis alert).
- You need a digital numeric pager.
- You need one of the following circuit packs:
  - — Call Classifier
  - — Tone-Clock (with Call Classification and Tone Detection)

Instructions

To set up crisis alert to a digital pager:

1. Type **change system-parameters crisis-alert** and press Return.

   The Crisis Alert System Parameters screen appears.

2. In the Alert Pager field, type **y**.

   This allows you to use the Crisis Alert to a Digital Pager feature and causes additional crisis alert administration fields to appear.

3. In the Originating Extension field, type a valid unused extension to send the crisis alert message.

   We’ll type **7768**.

4. In the Crisis Alert Code field, type 911.

   This is the number used to call the crisis alert pager.

5. In the Retries field, type **5**.

   This is the number of additional times the system tries to send out the alert message in case of an unsuccessful attempt.

6. In the Retry Interval (sec) field, type **30**.

   This is length of time between retries.

7. In the Main Number field, type the number that is to be displayed at the end of the pager message.

   We'll type **303-555-0800**.

8. In the Pager Number field, type the number for the pager.

   We'll type **303-555-9001**.

9. In the Pin Number field, type **pp77614567890**.

   This is the PIN number, if required, for the pager. Insert any pause digits (pp) as needed to wait for announcements from the pager service to complete before sending the PIN.

10. In the DTMF Duration - Tone (msec) field, type **100**.

    This is the length of time the DTMF tone is heard for each digit.

11. In the Pause (msec) field, type **100**.

    This is the length of time between DTMF tones for each digit.

12. Press ENTER to save your changes.

Related topics

Refer to ''Crisis Alert'' on page 1536 for more detailed information.

Other useful settings

There are many settings that control how your system operates and how your users phones work. Most of these you administer through one of the System Parameters screens. This section describes a few of the items you can enable in your system to help your users work more efficiently. See ''Feature-Related System Parameters'' on page 766 for a more detailed description of the available system settings.

Automatic callback if an extension is busy

You can allow users to request that the system call them back if they call a user whose telephone is busy. See ''Automatic Callback'' on page 1370.

Automatic hold

You can set a system-wide parameter that allows your users to initiate a call on a second line without putting the first call on Hold. This is called Automatic Hold, and you enable it on the Feature-Related System Parameters screen. If you do not turn this on, the active call drops when a the user presses the second line button.

Bridging onto a call that has gone to coverage

You can allow users to join (bridge) on to a call that rang at their extension and then went to coverage before they could answer. See ''Temporary Bridged Appearance'' on page 1747.

Distinctive ringing

You can establish different ringing patterns for different types of calls. For example, you can administer your system so that internal calls ring differently from external calls or priority calls. See ''Distinctive Ringing'' on page 1541 for more information.

Warning when phones are off-hook

You can administer the system so that if a phone remains off-hook for a given length of time, the switch sends out a warning. This is particularly useful in hospitals, where the phone being off-hook may be an indication of trouble with a patient. See ''Class of Service'' on page 650 for more information.

Warning users if their calls are redirected

You can warn analog phone users if they have features active that may redirect calls. For example, if the user has activated send all calls or call forwarding, you can administer the system to play a special dial tone when the user goes off-hook. See ''Special Dial Tone'' on page 791 for more information.

Managing phones

3

This section explains how to add, swap, upgrade, and remove different kinds of phones on your system. This section also gives you tips for customizing your own phone (for system administration) so it has the feature buttons you need for many administration and troubleshooting tasks. Most of the information applies to traditional DCP phones, and headings specifically state that certain procedures apply to IP (internet protocol) phones.

Note that this section does not tell you how to administer an attendant console. If you need to add or modify an attendant console, refer to ‘‘Managing your attendant consoles’’ on page 161.

Installing new phones

Simple administration allows you to plug a telephone into a jack and dial a sequence to start up service to the phone. The dialing sequence sets up an association between the phone and the corresponding station administration.

SECURITY ALERT:

If you do not manage this feature carefully, its unauthorized use may cause you security problems. Consult the Avaya Products Security Handbook for suggestions on how to secure your system and find out about obtaining additional security information.

For traditional instructions, see “Adding new phones.”

Before you start

On the Feature-Related System Parameters screen, be sure the Customer Telephone Activation (CTA) Enabled field is **y** and the TTI Enabled field is **y**.

Complete the station screen for the new phone and type x in the port field. Note that the phone type must match the board type. For example, match a two-wire digital phone with a port on a two-wire digital circuit pack. Use this procedure with all circuit-switched phones except BRI (ISDN) and model 7103A.

Tip:
See "Completing station screens" for more information. See Using templates if you want to add a number of phones with similar settings.

Instructions

CAUTION:
You can destroy your hardware if you attempt to connect an analog telephone to a digital port.

To associate a phone with existing x-port station administration, complete the following steps from the telephone you want to install:

1. Plug the phone into the wall jack.
2. Lift the receiver and continue if you hear dial tone.
3. Dial **#*nnnn**, where **nnnn** is the extension number of the phone you are installing.
4. Hang up after you receive confirmation tone.
5. Dial a test call to confirm that the phone is in service.

   If possible, call a phone with a display so the person answering can confirm that you entered the correct extension number.
6. Repeat the process until all new phones have been installed.
7. For security reasons, you should disable this feature when you are done. At the system administration terminal type **change system-parameters features** to access the Feature-Related System Parameters screen.
8. Type **n** in the Customer Telephone Activation (CTA) Enabled field.
9. Press ENTER to save your changes.
10. Type **save translations** and press ENTER to permanently save the changes.

    a. Fixing problems

If you misdial and the wrong extension is activated for the phone you are using, use the TTI unmerge feature access code to “uninstall” the phone before you try again. See “TTI separation from a telephone” for more information.

Adding new phones

When you are asked to add a new phone to the phone system, what do you do first? To connect a new phone you need to do three things:

- find an available port
- wire the port to the cross-connect field or termination closet
- tell the telephone system what you’re doing

Before you can determine which port to use for the new phone, you need to determine what type of phone you are installing, what ports are available, and where you want to install the phone.

To add an IP phone, see ‘‘Adding an IP Telephone’’ on page 104.

Gathering necessary information

1. Determine whether the phone is an analog, digital, ISDN, or hybrid set. You may also administer a virtual phone, one without hardware at the time of administration.

   You need this information to determine the type of port you need, because the port type and phone type must match. If you do not know what type of phone you have, refer to the Type field under ‘‘Station’’ on page 1066 for a list of phones by model number.

2. Record the room location, jack number, and wire number.

   You may find this information on the jack where you want to install the phone, recorded in your system records, or from the technician responsible for the physical installation.

3. Display the available boards (cards) and ports.

   To view a list of boards on your system, type **list configuration station** and press RETURN.

```
                         SYSTEM CONFIGURATION
Board                                                  Assigned Ports
Number     Board Type       Code    Vintage u=unassigned t=tti p=psa

01A05     DIGITAL LINE       TN754B 000002  01 u  03 u  05 u  07 08
01A06     ANALOG LINE        TN742  000010  01 02 03 04 u  u  u  u
01B05     ANALOG LINE        TN746B 000008  u  u  u  u  u  u  u  u
                                            u  u  u  u  u  u  u  u
01C04     ANALOG LINE        TN746B 000008  u  u  u  u  u  u  u  u
                                            u  u  u  u  u  u  u  u
01C05     DIGITAL LINE       TN2224 000004  01 u  u  04 u  u  07 08
                                            u  u  u  u  u  u  u  u
                                            u  u  u  u  u  u  u  u
01C06     HYBRID LINE        TN762B 000004  01 02 u  u  u  u  u  u
01C10     DIGITAL LINE       TN754  000004  u  u  u  u  u  u  u  u
```

The System Configuration screen shows all the boards on your system that are available for connecting phones. You can see the board number, board type, circuit-pack type, and status of each board's ports.

4. Choose an available port and record its port address.

   Each port that is available or unassigned is indicated by a '`u`.' Choose an available port from a board type that matches your phone type (such as a port on an analog board for an analog phone).

   Every phone must have a valid port assignment, also called a port address. The combined board number and port number is the port address. So, if you want to attach a phone to the 3rd port on the 01C05 board, the port address is 01C0503 (01=cabinet, C=carrier, 05=slot, 03=port).

   Tip:
   If you add several phones at one time, you may want to print a paper copy of the System Configuration screen. To print the screen to a printer attached to the system terminal, type **list configuration station print** *and press* RETURN. *To print to the system printer that you use for scheduled reports, type* **list configuration station schedule immediate** *and press* RETURN.

5. Choose an extension number for the new phone.

   The extension you choose must not be assigned and must conform to your dial plan. You should also determine whether this user needs an extension that can be directly dialed (DID) or reached via a central phone number.

   Be sure to note your port and extension selections on your system's paper records.

Physically connecting the phone

Once you have collected all the information, you are ready to physically wire the port to the cross-connect field.

If you have an Avaya representative or on-site technician who completes the physical connections, you need to notify them that you are ready to add the phone to the system. To request that Avaya install the new connections, call your Avaya representative to place an order.

If you are responsible for making the connections yourself and if you have any questions about connecting the port to the cross-connect field, refer to your system installation guide.

Now you are ready to configure the system so that it recognizes the new phone.

Before you start

To download foreign language display labels for phones, set the Display Language field on the Station screen to **english, spanish, italian, french**, or **user-defined**.

For a Eurofont character display for the 2420 phone, set the Display Character Set field on the System-Parameters Country-Options screen to **Eurofont**.

For a Katakana character display for the 2420 phone, set the Display Character Set field on the System-Parameters Country-Options screen to **Katakana**.

Completing the station screens

The information that you enter on the station screen advises the system that the phone exists and indicates which features you want to enable on the phone.

To access the station screen for the new phone:

1. Type **add station *nnnn*** and press RETURN, where *nnnn* is the extension for the new phone.

   Make sure the extension conforms to your dial plan. You can also use the **add station next** command to add a phone to the next available extension.

   When the Station screen appears, you see the extension number and some default field values. For example, the following screen is for a new phone at extension 2345.

```
                                STATION

Extension: 2345                       Lock Messages? n          BCC: _
     Type: 6480D+                     Security Code: _________   TN: 1_
     Port: _______________          Coverage Path 1: ____       COR: 1
     Name: _____________________    Coverage Path 2: ____       COS: 1
                                    Hunt-to Station: ________

STATION OPTIONS
          Loss Group: none___       Personalized Ringing Pattern: 1
         Data Module? n                         Message Lamp Ext: 2345
        Speakerphone: 2-way                   Mute Button Enabled? y
    Display Language: english
                                        MM Complex Data Ext:________
                                                   IP SoftPhone? n
```

2. Type the model number of the phone into the Type field.

   For example, to install a 8411D phone, type **6480D+** in the Type field. Note that the displayed fields may change depending on the model you add.

3. Type the port address in the Port field.
4. Type a name to associate with this phone in the Name field.

   The name you enter displays on called phones that have display capabilities. Also, some messaging applications, such as INTUITY, recommend that you enter the user's name (last name first) and their extension to identify the phone.

5. Press ENTER to save your changes.

To make changes to this new phone, such as assigning coverage paths or feature buttons, type **change station *nnnn*** and press RETURN, where *nnnn* is the extension of the new phone. Refer to ''Managing phone features'' on page 127 for more information.

Using templates to add phones

A quick way to add phones is to copy the information from an existing phone and modify it for each new phone. For example, you can configure one phone as a template for an entire work group. Then, you merely duplicate the template station screen to add all the other extensions in the group.

Note that only phones of the same model can be duplicated. The duplicate command copies all the feature settings from the template phone to the new phones.

To duplicate an existing phone:

1. Type **display station *nnnn*** and press RETURN.

   nnnn is the extension of the station screen you want to duplicate to use as a template. Verify that this extension is the one you want to duplicate.

2. Press CANCEL to return to the command prompt.

3. Type **duplicate station *nnnn*** and press RETURN, where *nnnn* is the extension you want to duplicate.

   The system displays a blank Duplicate Station screen.

```
                           STATION
                                 Security
Ext.     Port   Name                 Code      Room   Jack   Cable
_______  ______ ____________________ ________  ______ ______ ______
_______  ______ ____________________ ________  ______ ______ ______
_______  ______ ____________________ ________  ______ ______ ______
_______  ______ ____________________ ________  ______ ______ ______
_______  ______ ____________________ ________  ______ ______ ______
_______  ______ ____________________ ________  ______ ______ ______
_______  ______ ____________________ ________  ______ ______ ______
_______  ______ ____________________ ________  ______ ______ ______
_______  ______ ____________________ ________  ______ ______ _________
```

4. Type in the extension, port address, and phone name for each new phone you want to add.

   The rest of the fields are optional. You can complete them at any time.

5. Press ENTER to save your changes to system memory.

To make changes to these phones, such as assigning coverage paths or feature buttons, type **change station *nnnn*** and press ENTER, where *nnnn* is the extension of the phone that you want to modify.

Related topics

You can also add multiple call center agents, all with the same settings based on an agent that is already administered. Enter **command duplicate agent-loginID** and the extension of the agent you want to duplicate, then **start** and the extension you want to use for the first new agent, then **count** and the number of agents you want to add. Fill in the information on the Agent LoginID screen. For more information, refer to the *Avaya MultiVantage™ Call Center Guide to ACD Call Centers*.

Using an alias

Not every phone model or device has a unique station screen in the system. You might have to use an available model as an “alias” for another. If you need to enter a phone type that the system does not recognize or support, use an alias. Defining aliases is also a useful method to identify items that act as analog stations on the switch, such as a fax machine, a modem, or other analog device.

If you purchase a phone model that is newer than your system, you can alias this phone to an available model type that best matches the features of your new phone. Refer to your phone’s manual to determine which alias to use. If your manual does not have this information, you can contact the DEFINITY helpline for an appropriate alias.

For example, let’s create two aliases: one to add a new 6220 phone and one to add modems to our system.

1. Refer to your new phone’s manual to find the correct alias.

   In our example, we find that the 6220 should be administered on an older system as a 2500 phone.

2. Type **change alias station** and press RETURN.

   The Alias Station screen appears.

3. Enter **6220** in the Alias Set Type field.

   This is the name or model of the unsupported phone.

4. Enter **2500** in the Supported Set Type field.

   Enter the supported model in this field.

5. Enter **modem** in the second Alias Set Type field.

   You can call the alias set anything you like. Once you define the alias, you can use the alias set in the Type field on the Station screen.

6. Enter **2500** in the second Supported Set Type field.

   Entering 2500 indicates to the system that these models are basic analog devices.

7. Press ENTER to save your changes.

```
                  ALIAS STATION
  Alias Set Type               Supported Set Type
     6220                           2500
     modem                          2500
     _____                          _____
     _____                          _____
     _____                          _____
     _____                          _____
     _____                          _____
     _____                          _____
     _____                          _____
     _____                          _____
     _____                          _____
'#' indicates previously aliased set type is now native
```

Now you can follow the instructions for adding a new phone (or adding a fax or modem). Your switch now recognizes the new type (6220 or modem) that you enter in the Type field.

Be sure to refer to your phone's manual for instructions on how to set feature buttons and call appearance buttons. Note that if you need to use an alias for a phone, you may not be able to take advantage of all the features of the new phone.

Customizing your phone

This section provides recommendations for setting up or enhancing your personal phone. You need a phone that is powerful enough to allow you to use all the features you may give to other employees. You may want to add feature buttons that allow you to monitor or test the system, so that you can troubleshoot the system from your phone.

It will be much easier to monitor and test your system if you have a phone with:

- a large multi-button display (such as 8434D or 8410D)
- a class of service (cos) that has console permissions
- the following feature buttons
  - ACA and Security Violations (assign to lamp buttons)
  - Busy verify
  - Cover message retrieval button
  - Major/minor alarm buttons
  - Trunk ID buttons
  - Verify button

Once you select a phone, you'll want to determine if you want to place this phone at your desk or in the switch room. If the phone is in the switch room (near the system administration terminal), you can quickly add or remove feature buttons to test features and facilities. You may decide that you want a phone at both your desk and in the switch room — it's up to you.

You may also find it handy to set up multiple phones for testing applications and features before you provide them to users. You may want to have a phone that mimics each type of user phone in your organization. For example, if you have four basic phone templates, one for executives, one for marketing, one for technicians, and one for other employees, you may want to have examples of each of these phones so you can test new features or options. Once you are satisfied that a change works on the test phone, you can make the change for all the users in that group.

Upgrading phones

If you want to change phone types for a user and do not need to change locations, you can just access the station screen for that extension and enter the new model number.

Tip:
This method can be used only if the new phone type matches the existing port type (such as digital phone with a digital port).

For example, if a user at extension 4556 currently has a 7410+ phone and you want to replace it with a new 8411D phone:

1. Type **change station 4556** and press RETURN.

   The Station screen for 4556 appears.

2. Overwrite `7410+` with **8411D** in the Type field.

   Now you can access the functions and feature buttons that correspond to an 8411D phone.

Swapping phones

You will often find that you need to move or swap phones. For example, employees moving from one office to another may want to bring their phones. In this case, you can use X ports to easily swap the phones.

In general, to swap one phone (phone A) with another phone (B), you change phone A's port assignment to **x**, change phone B's port assignment to A's old port, and, finally, change the **x** for phone A to B's old port. Note that these swapping instructions work only if the two phones are the same type (both digital or both analog, etc.).

For example, to swap phones for extension 4567 (port 01C0505) and extension 4575 (port 01C0516), complete the following steps:

1. Type **change station 4567** and press RETURN.
2. Record the current port address (`01C0505`) and type **x** in the Port field.
3. Press ENTER to save your changes.
4. Type **change station 4575** and press RETURN.
5. Record the current port address (`01C0516`).
6. Type **01C0505** in the Port field.
7. Update the Room and Jack fields.
8. Press ENTER to save your changes.
9. Type **change station 4567** again and press RETURN.
10. Type **01C0516** in the Port field.

    This is the port that used to be assigned to extension 4575.
11. Update the Room and Jack fields.
12. Press ENTER to save your changes.
13. Physically unplug the phones and move them to their new locations.

When you swap phones, the system keeps the old button assignments. If you are swapping to a phone with softkeys, the phone could have duplicate button assignments, because softkeys have default assignments. You may want to check your button assignments and modify them as necessary.

Using ACTR to move phones

Automatic Customer Telephone Rearrangement (ACTR) allows a phone to be unplugged from one location and moved to a new location without additional switch administration. The switch automatically associates the extension to the new port. ACTR works with 6400 Serialized phones and to the 2420 phones. The 6400 Serialized phone is stamped with the word “Serialized” on the faceplate for easy identification. The 6400 Serialized phone memory electronically stores its own part ID (comcode) and serial number as does the 2420 phone. ACTR uses the stored information and associates the phone with new port when the phone is moved.

ACTR is an enhancement to Terminal Translation Initialization (TTI), Personal Station Access (PSA), Customer Telephone Activation (CTA). ACTR makes it easy to identify and move phones.

CAUTION:

When a phone is unplugged and moved to another physical location, the Emergency Location Extension field must be changed for that extension or the USA Automatic Location Identification data base must be manually updated. If the Emergency Location Extension field is not changed or if the USA Automatic Location Identification data base is not updated, the DID number sent to the Public Safety Network could send emergency response personnel to the wrong location.

Detailed description

On the Feature-Related System Parameters screen, set the Terminal Translation Initialization (TTI) Enabled field to **y** and the TTI State field to **voice**.

NOTE:

When a phone is moved, if there is any local auxiliary power (a power supply plugged into a local AC outlet), the phone must be plugged into an AC outlet at the phone’s new location. A phone with remote auxiliary power must be supplied remote auxiliary power at its new location. If you do not supply auxiliary power in either case after a phone is moved, some optional adjuncts (for example, an expansion module) do not operate.

When you enter **always** or **once** in the Automatic Moves field on the station screen, the switch adds the extension to its ACTR Move List database. When the phone is plugged in, the switch asks the phone for its serial number and records the serial number on the ACTR Move List. If you change the entry in the Automatic Moves field from **always** or **once** to **no**, the switch removes the extension from the Move List.

Call processing

When a phone is unplugged while on a call, and a 6400 Serialized phone or a 2420 phone that is administered for automatic moves is plugged into the port within 60 seconds:

- both extensions are placed in idle state
- active calls on either extension are dropped, unless the call is active on a bridged appearance at some other phone
- held calls remain in a hold state
- any calls ringing on either extension instantly proceed to the next point in coverage or station hunting path, unless the call is ringing on a bridged appearance at some other phone
- user actions that were pending when the new phone was plugged in are aborted

Design considerations

You can use the **list station movable** command to keep track of extensions on the move list. Once you reach the maximum number, the switch does not allow additional extensions.

Administration

Before you start

- Be sure the TTI field on the Feature-Related System Parameters screen is set to **y**.

Tasks

Before you move a phone in your system, set the TTI State field to voice on the Feature-Related System Parameters screen.

Moving phones

You can allow a phone to be unplugged from one location and moved to a new location without additional switch administration.

For example, to allow moves anytime for a phone at extension 1234:

1. Type **change station 1234** and press RETURN.
2. Move to the Automatic Moves field.

```
change station 1014                                           Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? msa-spe      Auto Select Any Idle Appearance? n
          LWC Activation? y                   Coverage Msg Retrieval? y
  LWC Log External Calls? n                              Auto Answer: none
            CDR Privacy? n                          Data Restriction? n
   Redirect Notification? y               Idle Appearance Preference? n
 Per Button Ring Control? n
   Bridged Call Alerting? n                 Restrict Last Appearance? y
  Active Station Ringing: single

       H.320 Conversion? n          Per Station CPN - Send Calling Number? _
      Service Link Mode: as-needed  Special Character for Restricted Number? n
        Multimedia Mode: basic
    MWI Served User Type: ______              Display Client Redirection? n
        Automatic Moves: always
             AUDIX Name:                     Select Last Used Appearance? n
   Messaging Server Name: ______               Coverage After Forwarding? _
     Recall Rotary Digit? n                      Multimedia Early Answer? n
                                          Direct IP-IP Audio Connections? n
                                                    IP Audio Hairpinning? n
```

3. Type **always** in the Automatic Moves field.
4. Press ENTER to save your changes.

Using TTI to move phones

Terminal Translation Initialization (TTI) allows you to merge an x-ported station to a valid port by dialing a TTI merge code, a system-wide security code, and the x-port extension from a telephone connected to that port. TTI also allows you to separate an extension from its port by dialing a similar separate digit sequence. This action causes the station to revert to an x-port.

TTI can be used for implementing telephone and data module moves from office to office. That is, you can separate a telephone from its port with TTI, unplug the telephone from the jack, plug in the telephone in a jack in a different office, and merge the telephone to its new port with TTI.

If you are moving phones and concerned about security, you may also want to refer to “Setting up Personal Station Access’’ on page 354 for more information about setting the security code for each extension.

SECURITY ALERT:

If you do not manage this feature carefully, its unauthorized use may cause you security problems. For example, someone who knows the TTI security code could disrupt normal business functions by separating telephones or data terminals. You can help protect against this action by frequently changing the TTI security code. You can further enhance system security by *removing the FAC from the system when it does not need to be used (for example, there are no moves going on at present). Consult the Avaya Products Security Handbook for additional steps to secure your system and find out about obtaining information regularly about security developments.*

Before you start

Before you can merge a telephone, you must set the TTI State field to **voice** on the Feature-Related System-Parameters screen. You also must set the extension to match the port type of the TTI port making the merge request. For example, a digital telephone type can merge only to a port on a digital board.

NOTE:
If you use CAMA trunks and you are moving a phone to a different physical location, be sure to update your emergency service provider’s database. Since emergency teams use this database to determine the phone’s location during an emergency call, you should help keep this database current.

NOTE:
You cannot use TTI to change a virtual extension.

Instructions

TTI merge from a voice TTI port

CAUTION:
You can destroy your hardware if you attempt to connect an analog telephone to a digital port.

To merge an extension to a telephone with TTI, complete the following steps from the telephone you want to merge:

1. Dial the TTI merge FAC.
   - If the code is correct, you receive dial tone.
   - If the code is not correct, you receive intercept tone.
2. Dial the TTI security code from the telephone you want to merge.
   - If the code is correct, you receive dial tone.
   - If the code is not correct, you receive intercept tone.

3. Dial the extension of the telephone you want to merge.
   — If the extension is valid, you receive confirmation tone, which may be followed by dial tone. (It is possible to receive intercept tone immediately following the confirmation tone. If this happens, you need to attempt the merge again.)
   — If the extension is valid, but the extension is being administered, you receive reorder tone. Try the merge again later.
   — If the extension is invalid, you receive intercept tone.
   — If the system is busy and cannot complete the merge, you receive reorder tone. Try the merge again later.
   — If the telephone has a download status of pending, you receive reorder tone. You need to change the download status to complete to successfully complete the TTI merge.

TTI separation from a telephone

To complete a TTI separation, complete the following steps from the telephone that needs to be separated:

1. Dial the TTI separate FAC.
2. Dial the TTI security code.
   — If the code is correct, you receive dial tone.
   — If the code is not correct, you receive intercept tone.
3. Dial the extension of the telephone to be separated.
   — If you have dialed the extension of the telephone currently merged with this telephone, you receive confirmation tone.
   — If you have dialed the extension of the telephone currently merged with this telephone, but the extension is being administered, you receive reorder tone. Try the separation again later.
   — If you have not dialed the extension of the telephone currently merged with this telephone, you receive intercept tone.
   — If the system is busy and cannot complete the separation, you receive reorder tone. Try the separation again later.

Fixing problems

If you are having difficulty using TTI, you may want to review the following system restrictions:

- The TTI Ports field on the System Capacity screen shows the number of TTI ports used in a switch. This field shows only the number of TTI ports being administered. If a TTI exceeds the maximum number of ports, the port is not administered and cannot be added. In that case, a telephone cannot be added.

  BRI endpoints are only counted as one TTI port. For example, for every two BRI endpoints, one TTI port is counted. As such, you can have two telephones assigned to one port. If either endpoint is administered, the TTI port count is reduced by 1.

- The total number of translated telephones and Voice TTI ports in a system is limited to the maximum number of administered telephones supported in the system. The total number of translated data terminals and Data TTI ports in a system is limited to the maximum number of administered data modules allowed in the system.

- Set the TTI State field to **voice** and then set the TTI State field to **data**. When you use this order, voice and then data, you reduce the chance of a user trying to use TTI on a data-only terminal that does not have TTI port translation. This can happen when the number of telephones allowed by the system is twice the number of data terminals. For example, if the system limit for telephones is 15,000 and 7,500 for data, then when TTI was turned on for data first, only the first 7,500 unadministered ports would get TTI port translations.

- When TTI is activated for the system, the following actions take place:
  - If the TTI State field was previously activated but in a different state (such as, a voice to data state), the old TTI translations are removed and the new ones added on a board by board basis.
  - If the TTI State field is set to **voice**, then default TTI translations are generated for every unadministered port on all digital, hybrid, and analog boards.
  - If the TTI State field is set to **data**, then default TTI translations are generated for every unadministered port on all digital and data line boards in the system.
  - Whenever a new digital board is inserted when the system is in TTI Data mode, or when a digital, hybrid, or analog board is inserted when the system is in TTI Voice mode, the unadministered ports on the board become TTI ports.
  - When TTI is deactivated, all translation for the TTI ports are removed in the system; the ports return to an unadministered state.

Removing phones

Before you physically remove a phone from your system, check the phone's status, remove it from any group or usage lists, and then delete it from the system's memory.

For example, to remove a phone at extension 1234:

1. Type **status station 1234** and press RETURN.

   The General Status screen appears.

2. Make sure that the phone:

   - is plugged into the jack
   - is idle (not making or receiving calls)
   - has no messages waiting
   - has no active buttons (such as Send All Calls or Call Forwarding)

3. Type **list groups-of-extension 1234** and press RETURN.

   The Extension Group Membership screen shows whether the extension is a member of any groups on the system.

4. Press CANCEL.

5. If the extension belongs to a group, access the group screen and delete the extension from that group.

   For example, if extension 1234 belongs to pickup group 2, type **change pickup group 2** and delete the extension from the list.

6. Type **list usage extension 1234** and press RETURN.

   The Usage screen shows where the extension is used in the system.

7. Press CANCEL.

8. If the extension appears on the Usage screen, access the appropriate feature screen and delete the extension.

   For example, if extension 1234 is bridged onto extension 1235, type **change station 1235** and remove the appearances of 1234.

9. Type **change station 1234** and press RETURN.

10. Delete any bridged appearances or personal abbreviated dialing entries and press ENTER.

11. Type **remove station 1234** and press RETURN.

    The system displays the station screen for this phone so you can verify that you are removing the correct phone.

    Tip:
    Be sure to record the port assignment for this jack in case you want to use it again later.

12. If this is the correct phone, press ENTER.

    If the system responds with an error message, the phone is busy or still belongs to a group. Press CANCEL to stop the request, correct the problem, and enter **remove station 1234** again.

13. Remove the extension from voice mail service if the extension has a voice mailbox.

14. Type **save translations** and press RETURN to save your changes.

Note that you do not need to delete the extension from coverage paths. The system automatically adjusts coverage paths to eliminate the extension.

Now you can unplug the set from the jack and store it for future use. You do not need to disconnect the wiring at the cross-connect field. The extension and port address remain available for assignment at a later date.

Once you successfully remove a set, that set is permanently erased from system memory. If you want to reactivate the set, you have to add it again as though it were a new phone.

Adding a fax or modem

Connecting a fax machine or modem to your system is similar to adding a phone, with a few important exceptions. If you have not added a phone, you may want to read ''Adding new phones'' on page 87.

Because the system does recognize the concept of "fax" or "modem," you need to administer these items as basic analog stations. You can merely use the supported station type 2500 (analog, single line).

Alternatively, you can create aliases to the 2500 for fax machines and modems. If you want to be able to create reports that indicate which stations are faxes or modem, you should create aliases for these items. For more information about aliasing, refer to ''Using an alias'' on page 92.

For this example, let's assume that we have already defined an alias for 'fax' as a 2500 and that we now want to add a fax machine to extension 4444.

To add a fax machine as extension 444, complete the following steps:

1. Type **add station 4444** and press RETURN.
2. In the Type field, type **fax**.
3. In the Port field, type the port address.
4. In the Name field, type a name to associate with this fax.
5. Move to the Data Restriction field and type **y**.

   Entering y in this field prevents calls to and from this extension from being interrupted by tone signals. This is important for fax machines and modems as these signals can disrupt transmissions of data.
6. In the Distinctive Audible Alert field, type **n**.

   This eliminates the distinct 2-burst ring for external calls, which often interferes with the auto-answer function on fax machines or modems.
7. Press ENTER to save your changes.

Adding an IP Telephone

The 4600-series IP Telephones are physical sets that connect to the MultiVantage system via TCP/IP.

CAUTION:

An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It only reaches the local emergency service in the Public Safety Answering Point area where the telephone system is located. Please be advised that an Avaya IP endpoint cannot dial to and connect with local emergency service when dialing from remote locations. You should not use an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations.

Before you start

Be sure that your system has been enabled to use IP Telephones. Display the System-Parameters Customer-Options screen and verify the following field settings:

- Maximum Concurrently Registered IP Stations is greater than **0**
- IP Stations field is **y**
- Information has been entered in the fields on the Maximum IP Registrations by Product ID page

Instructions for adding an IP Telephone

Let's add an IP telephone at extension 4005.

To assign an extension, complete the following steps:

1. Type **add station 4005** and press RETURN.

   The Station screen appears.

```
                              STATION

Extension: 2345                    Lock Messages? n          BCC: _
     Type: 4620+                   Security Code: _________   TN: 1_
     Port: IP                    Coverage Path 1: ____       COR: 1
     Name: _____________________ Coverage Path 2: ____       COS: 1
                                 Hunt-to Station: _____

STATION OPTIONS
          Loss Group: none___    Personalized Ringing Pattern: 1
                                             Message Lamp Ext: 2345
        Speakerphone: 2-way               Mute Button Enabled? y
    Display Language: english                Expansion Module? y

                                          MM Complex Data Ext:
                                                 IP SoftPhone? n
```

   In the Type field, type **4620**.

2. The Port field is display-only, and **IP** appears.
3. In the Security Code field, enter a password for the IP phone user.

   NOTE:
   Although the system accepts a null password, the IP phone will not work unless you assign a password.

4. Press ENTER to save your work.

Changing from dual-connect to single-connect IP phones

When you have a dual extension phone and you upgrade to a single extension phone, you can remove the connection that is no longer used for that phone.

To remove the H.323 connection that is no longer needed, first record the media complex extension number:

1. Type **change station *nnnn***, where nnnn is the extension number of the original dual-connect phone that you are replacing with a single-connect phone.

   The Station screen appears.

2. Move to the Media Complex Extension field.
3. Write down the number in the Media Complex field, then delete the number from the field.
4. Press ENTER to save your changes.

Now remove the extension you recorded. Before you remove an H.323 extension from your system, check the status, remove it from any group or usage lists, and then delete it from the system's memory.

For example, if you wrote down extension 1234 before you removed it from the Media Complex field on the Station screen, then remove extension 1234 using these steps:

1. Type **status station** 1234 and press RETURN.

   The General Status screen appears.

2. Make sure that the extension:
   - is idle (not making or receiving calls)
   - has no messages waiting
   - has no active buttons (such as Send All Calls or Call Forwarding)
3. Type **list groups-of-extension 1234** and press RETURN.

   The Extension Group Membership screen shows whether the extension is a member of any groups on the system.

4. Press CANCEL.
5. If the extension belongs to a group, access the group screen and delete the extension from that group.

   For example, if extension 1234 belongs to pickup group 2, type **change pickup group 2** and delete the extension from the list.

6. Type **list usage extension 1234** and press RETURN.

   The Usage screen shows where the extension is used in the system.

7. Press CANCEL.

8. If the extension appears on the Usage screen, access the appropriate feature screen and delete the extension.

   For example, if extension 1234 belongs to hunt group 2, type **change hunt group 2** and delete the extension from the list.

9. Type **change station 1234** and press RETURN.

10. Delete any bridged appearances or personal abbreviated dialing entries and press ENTER.

11. Type **remove station 1234** and press RETURN.

    The system shows the station screen for this phone so you can verify that you are removing the correct phone.

12. If this is the correct phone, press ENTER.

    The system responds with **command successfully completed**.

    If the system responds with an error message, the phone is busy or still belongs to a group. Press CANCEL to stop the request, correct the problem, and enter **remove station 1234** again.

13. Remove the extension from voice mail service if the extension has a voice mailbox.

14. Type **save translations** and press RETURN to save your changes.

Note that you do not need to delete the extension from coverage paths. The system automatically adjusts coverage paths to eliminate the extension.

Once you successfully remove the extension, it is permanently erased from system memory. If you want to reactivate the extension, you have to add it again as though it were new.

Tasks

Setting up emergency calls on IP phones

Set up which “calling number” to send to the public safety access point when an emergency call is placed from an IP phone.

Instructions

You use the Station screen to set up emergency call handling options for IP phones. As an example, we'll administer the option that prevents emergency calls from an IP phone.

To prevent an emergency call from an IP phone:

1. Type **change station *nnnn*** and press enter, where ***nnnn*** is the extension of the phone you want to modify.

   The station screen appears.

```
change station 1014                                                  Page 1 of X
                              STATION

Extension: 1014               Lock Messages? n                         BCC:
     Type:                    Security Code:                            TN:1
     Port:                  Coverage Path 1:                           COR: 1
     Name:                  Coverage Path 2:


STATION OPTIONS
              Loss Group: 2            Personalized Ringing Pattern: 3
             Data Module? n                        Message Lamp Ext: 1014
            Speakerphone: 2-way               Mute button enabled? y
        Display Language? English                 Media Complex Ext:
                                                       IP Softphone? y
                                             Remote Office Station? n
                                                IP Emergency calls: block
```

2. Type **block** in the IP Emergency calls field and press RETURN to save your changes.

Fixing problems

Problem	Possible causes	Solutions
Audio levels can not be adjusted.	The TN802B Medpro circuit pack is being used.	Use TN799B or later C-LAN circuit pack and the TN2302 IP Media Processor

Adding an IP Softphone

Avaya IP Softphones enable the end user to control telephone calls directly from a personal computer (PC). An end user can log into your company's MultiVantage server remotely and make and receive telephone calls from the telephone extension.

Avaya IP Softphone supports the following two configurations:

- road-warrior application

  You typically use this configuration for laptop users who are travelling. In this configuration, the PC LAN connection carries both the call control signaling and the voice path. Because voice calls are routed over the connection to the PC, you'll need a headset connected to the PC or a handset that plugs into the PC's sound card, to speak and hear sounds.

- telecommuter application or Avaya IP Agent

  For the telecommuter or Avaya IP Agent configuration, you make two separate connections to the DEFINITY server. The signaling path is carried over an IP network and the voice path is carried over the standard circuit-switched telephone network (PSTN). Since you are using a phone for audio, you do not need an H.323 PC audio application.

  The telecommuter configuration uses the Avaya IP Softphone interface (on the user's PC) and a standard phone. The Avaya IP Agent configuration uses the Avaya IP Agent interface (on the agent's PC) and a call center phone, such as a CallMaster.

Tip:
Use status station to show the part (product) ID, serial number and the audio connection method used by existing stations.

Before you start

Be sure that your system has been enabled to use IP Softphones. Display the System Parameters Customer-Options screen and verify the following field settings:

- Maximum Concurrently Registered IP Stations is greater than 0
- IP Stations field is **y**
- Information has been entered in the fields on the Maximum IP Registrations by Product ID page

Be sure that your system has a CLAN board and an IP Media Processor board.

Once you're finished administering your MultiVantage system, you need to install the IP Softphone software on each user's PC.

Instructions for adding a road-warrior application

You can use the road-warrior application when you have only a single telephone line available to access the MultiVantage system over the IP network.

You also can “take over” an IP phone. Typically you would not have a different extension for your softphone. When you log in, the softphone takes over the existing phone extension (turn the DCP or IP phone off). During this time, that DCP or IP phone is out of service. This is accomplished if, on the Station screen, the IP Softphone field is **y**.

Let’s add a road-warrior application at extension 3001. Except for single-connect IP phones, you have to actually administer two extensions for each road-warrior application. We will first add an H.323 extension at 3000.

To assign an H.323 extension, complete the following steps:

1. Type **add station 3000** and press RETURN.

   The Station screen appears.

```
add station 3000                                                  Page   1 of 3
                                   STATION

Extension: 3000                        Lock Messages? n              BCC: 0
     Type: H.323                       Security Code:                 TN: 1
     Port: IP                        Coverage Path 1:                COR: 1
     Name:                           Coverage Path 2:                COS: 1
                                     Hunt-to Station:              Tests? y

STATION OPTIONS
              Loss Group: 2
```

2. In the Type field, enter **H.323**.
3. Press ENTER to save your work.

Now, you need to administer the phone (DCP) extension. To do so, complete the following steps:

1. Type **add station 3001** and press RETURN.

   The Station screen appears. Note that you choose to change an existing DCP extension by using **change station *nnnn*** in this step, where *nnnn* is the existing DCP extension.

```
add station 3001                                              Page   1 of   4
                                  STATION

Extension: 3001                        Lock Messages? n            BCC: 0
     Type: 6408D                       Security Code: *             TN: 1
     Port: X                         Coverage Path 1:              COR: 1
     Name:                           Coverage Path 2:              COS: 1
                                     Hunt-to Station:

STATION OPTIONS
              Loss Group: 2         Personalized Ringing Pattern: 1
             Data Module? n                    Message Lamp Ext: 3000
            Speakerphone: 2-way             Mute Button Enabled? y
        Display Language: english

                                                 Media Complex Ext: 3000
                                                          IP Softphone? y
```

2. In the Type field, enter the model of phone you want to use, such as **6408D**.
3. In the Port field, type **x** for virtual phone or enter the port number if there is hardware.
4. In the Security Code field, enter the password for this remote user, such as **1234321**.

   This password can be up to 7 digits in length.
5. In the Media Complex Ext field, type **3000**.

   This is the H.323 extension we just administered.
6. In the IP Softphone field, type **y.**
7. On page 2, in the Service Link Mode field, type **as-needed**.

   Set this field to **permanent** only for extremely busy remote phone users, such as call center agents.
8. In the Multimedia Mode field, type **enhanced**.
9. Press ENTER to save your work.

Now you can install and configure the software on the user’s PC. In this example, the user will login by entering their DCP extension (3001) and password (1234321).

Fixing problems

Problem	Possible causes	Solutions
Display characters on the phone can not be recognized.	Microsoft Windows is not set to use Eurofont characters.	Set the Microsoft Windows operating system to use Eurofont.

Instructions for adding a telecommuter application

Assign this configuration to remote users who have two available phone lines. For example, to administer a telecommuter application for a home user at extension 3010, complete the following steps:

1. Type **add station** 3010 and press RETURN.

   The Station screen appears.

   NOTE:
   Use the **add station** command if this is a new DCP extension. Use the **change station** command for an existing DCP extension and ignore steps 2 and 3.)

2. In the Port field, type **x** for virtual phone or enter the port number if there is hardware.
3. In the Security Code field, enter the password for this remote user, such as **1234321**.

   This password can be up to 7 digits in length.

4. In the IP Softphone field, type **y**.
5. On page 2, in the Service Link Mode field, type **as-needed**.

   Set this field to **permanent** only for extremely busy remote phone users, such as call center agents.

6. In the Multimedia Mode field, type **enhanced**.
7. Press ENTER to save your work.

Now you can install and configure the software on the user's PC. In this example, the user will login by entering their DCP extension (3010) and password (1234321).

Related topics

Refer to the ''Internet Protocol (IP) Softphones'' on page 1329 for descriptions of the IP Softphone configurations.

Refer to the online help and to *Avaya IP Softphone Overview and Troubleshooting* for customer information on Avaya IP Softphone applications. This document is a Portable Document Format (PDF) document that is located in the Overview Document folder on the Avaya IP Softphone CD.

Also refer to *Getting Started*, located on the Avaya IP Softphone CD for more information on how to install and configure the IP Softphone software.

Setting up remote office

MultiVantage remote office provides IP processing capabilities to traditional call handling for voice and data between MultiVantage and offices with remote office hardware. You need to add the remote office information as a node on the MultiVantage, add extensions, set up the trunk and signaling groups.

Before you start

Be sure the following fields on the Feature Related System Parameters Customer Options screen are set to **y** or completed. If not, contact your Avaya representative.

- Maximum Administered Remote Office Trunks
- Maximum Administered Remote Office Stations
- Product ID registration limit
- Remote Office
- IP station
- ISDN-PRI
- IP trunks
- MMCH Basic and enhanced for trunks

Also, be sure your remote office hardware is installed and administered at the remote location. You need the following information from the remote administration:

- IP address
- Password

Adding remote office to Avaya MultiVantage Solution

In our example, we'll set up a remote office location using Avaya R300 Remote Office Communicator hardware in our branch office in Santa Fe. We'll add a new node, and set up the signaling group and trunk group.

Adding a node

Instructions

To add the remote office node to MULTIVANTAGE:

1. Type **change node-names IP** and press RETURN.

   The Node Name screen appears.

```
change node-names ip                                                 Page 1 of 1
                               IP NODE NAMES

  Name                  IP Address               Name          IP Address
default               0   .0   .0   .0       __________________  ___.___.___.___
remote office 1       134.23.107.22          __________________  ___.___.___.___
__________________  ___.___.___.___      __________________  ___.___.___.___
__________________  ___.___.___.___      __________________  ___.___.___.___
__________________  ___.___.___.___      __________________  ___.___.___.___
__________________  ___.___.___.___      __________________  ___.___.___.___
__________________  ___.___.___.___      __________________  ___.___.___.___
__________________  ___.___.___.___      __________________  ___.___.___.___
__________________  ___.___.___.___      __________________  ___.___.___.___
__________________  ___.___.___.___      __________________  ___.___.___.___
```

Screen 1. IP Node Names

2. In the Name field, type in a word to identify the node.

   In our example, type **Remote 6**.

3. In the IP address field, type in the IP address to match the one on the Avaya R300 administration.
4. Press ENTER to save your changes.

5. Type **add remote office** and the number for this remote office, and press RETURN.

   The remote office screen appears.

```
add remote-office 6                                                Page 1 of 1
                              REMOTE OFFICE 6

     Node Name:  Remote Office 6
Network Region:  22
      Location:  1
     Site Data:  Contact: Joe Smith
                 Phone: xxx-yyy-zzz
                 _______________________________
```

Screen 2. Remote Office

6. Fill in the following fields:
   - Node Name - match the name on the IP Node Names screen.
   - Network Region - this must match the network region on the IP-Interface screen for the circuit packs that connect this remote office. Use **display ip-interfaces** to find this information.
   - Location - match the one set up on the Location screen for this remote office.
   - Site Data - identify the street address or identifier you want to use
7. Press ENTER to save your changes.

Tip:
Use **status remote office** *to verify that your switch recognizes the remote office information. It also displays the extensions and signaling group you administer next.*

Setting up a trunk group

You can modify an existing trunk group or add a new one. In our example, we'll add trunk group 6.

Before you start

Perform "Setting up a signaling group'' on page 116.

To set up the trunk group for your remote office:

1. Type **add trunk group 6**.

   The Trunk Group screen appears.

```
add trunk-group next                                                  Page 1 of x
                                 TRUNK GROUP

 Group Number: 6                    Group Type: ISDN          CDR Reports: _
   Group Name: Remote office 6             COR: __       TN: ___      TAC: 6
    Direction: ________         Outgoing Display? _        Carrier Medium: IP
  Dial Access? _                 Busy Threshold: ___     Night Service: ________
 Queue Length: ___                              Incoming Destination: ________
    Comm Type: _____                 Auth Code? _
                                   Trunk Flash? _
          BCC: _                           ITC? ____
 TRUNK PARAMETERS
    Trunk Type (in/out): ____________    Incoming Rotary Timeout(sec): __
     Outgoing Dial Type: _________                 Incoming Dial Type: ________
                                              Disconnect Timing(msec): ____
        Digit Treatment: __________                            Digits: ____
                                                    Sig Bit Inversion: none
      Analog Loss Group: ___                       Digital Loss Group: ___
     Incoming Dial Tone? _
               Bit Rate: _____     Synchronization: _____      Duplex: ____
  Disconnect Supervision - In? _  Out? _
  Answer Supervision Timeout: ___          Receive Answer Supervision? _
```

2. In the Group type field, type ISDN.

   ISDN-PRI or ISDN-BRI must be y on the System Parameters Customer Options screen.

3. In the TAC field, type in the trunk access code that conforms to your dial plan.
4. In the Carrier Medium field, type **IP** (Medpro).
5. In the Dial Access field, type **y**.
6. In the Service type field, type **tie**.
7. In the Signaling Group field, type in the signaling group you created.
8. Press ENTER to save your changes.

Setting up a signaling group

Each Remote Office has own listen port and signaling group. Set up a new trunk group, or use an existing trunk group administered for H.323 signaling.

Instructions

Set up the signaling group for remote office:

1. Type **add signaling-group** and the number of the group you want to add.

   The signaling group screen appears.

```
add signaling-group 6                                              Page 1 of 5
                               SIGNALING GROUP

Group Number  6            Group Type:  H.323
                         Remote Office?  y     Max Number of NCA TSC: ___

                                                Max number of CA TSC:  ___
                                             Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  6
        Supplementary Service Protocol: _


  Near-end Node Name:  clan              Far-end Node Name: remote office 6
Near-end Listen Port:  5001            Far-end Listen Port:5055

        LRQ Required? _          Calls Share IP Signaling Connection?  _
        RRQ Required? y
                                     Bypass If IP Threshold Exceeded?  _

                                         Interworking Message: ________
```

2. In the Group Type field, type **H.323**.
3. In the Remote Office field, type **y**.
4. In the Trunk Group for Channel Selection, type the number of the trunk you set up for the remote office.
5. In the Near-end Node Name, identify the node name assigned to the CLAN that supports the R300.
6. In the Far-end Node Name, identify the node name assigned to the CLAN that supports the R300.
7. In the Near-end Listen port, type a port number in the 5000-9999 range.
8. In the Far-end Listen Port field, type **1720**. See *Getting Started with the Avaya R300 Remote Office Communicator* for more details.
9. In the RRQ field, type **y**.
10. Tab to the Direct IP-IP Audio Connection field on another page of this screen and type **y**.
11. Press ENTER to save your changes.

Setting up remote office on network regions

Now we will set up a network region and show the connections between regions.

Instructions

Set up network region 1:

1. Type **change ip-network-region 1** and press ENTER.

   The IP Network Region screen appears.

```
add ip-network-region 1                                           Page 1 of 2
                               IP Network Region

                     Region: 1
                       Name: Denver main
          Audio Parameters
                  Codec Set: 1

             UDP Port Range
                        Min: _____
                        Max: _____
      Diff Serve PHB Value: ___              Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? _
           802.1p.Q Enabled? _
```

2. In the Name field, describe the region you are setting up.
3. In the Codec-set field, type the codec-set you want to use in this region.
4. In the UDP Port Range, type the range of the UDP port number to be used for audio transport.
5. In the Direct IP-IP Audio Connections field, type y.
6. Move to page 2 to set up connections between regions and assign codecs for inter-region connections.

```
add ip-network-region 1                                         Page 2 of 2
                      Inter Network Region Connection Management

Region                                (Group of 32)
        1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2
001-032 1 _ _ 2 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
033-064 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
065-096 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
097-128 _ _ 5 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
129-160 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
161-192 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
193-224 6 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
225-250 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
```

The following connections are administered in this example:

- codec-set 2 is used between region 1 and region 4
- codec-set 5 is used between region 1 and region 99
- codec-set 6 is used between region 1 and region 193.

Now you need to assign the region number to the CLAN circuit pack. All the endpoints registered with a specific CLAN circuit pack belong to the CLAN’s region. See *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

Adding phones to remote office

Be sure the extensions you add fit your dialing plan.

Instructions

1. Type **add station nnnn** and press ENTER, where nnnn is the extension you are adding.

   The station screen appears.

```
add station 6001                                                 Page 1 of X
                                          STATION

Extension: 1014                              Lock Messages? n         BCC: 0
     Type: 8410D                             Security Code: 1234567    TN: 1
     Port: x                                 Coverage Path 1: ___     COR: 1
     Name: Remote main                       Coverage Path 2: ___     COS: 1
                                              Map-to Station:
                                             Hunt-to-Station: _______

STATION OPTIONS
              Loss Group: _               Personalized Ringing Pattern:
             Data Module? n                         Message Lamp Ext: 6001
            Speakerphone: 2-way                  Mute button enabled? y
        Display Language? English
                                                   Media Complex Ext:
                                                IP Station? n
                                                   Remote Office Phone? y
```

2. In the Type field, type in the model of the phone you are adding.
3. In the Port field, type **x**.
4. In the Name field, identify the phone for your records.

5. In the Security Code field, match the password set up on the Remote Office administration.
6. In the Remote Office field, type **y**.
7. In the Direct IP-IP Audio Connection field, type **y**.
8. Press ENTER to save your changes.

Tip:
You can set up a telnet session on your remote office administration program to verify that the phone is registered. See *Getting Started with the Avaya R300 Remote Office Communicator* for *details.*

Location by region

Location by Region provides a way to administer location by IP network region. This allows for the correct date and time information and trunk routing based on the IP network region.

Before you start

Be sure the following fields on the System Parameters Customer-Options screen are set to **y**.

- IP Stations
- IP Trunks
- Remote Office

Setting up a remote user in a network region that is correct for their time zone

Instructions

With your system located in New York and a remote user located in Germany, to create the correct time zone settings:

1. Type **change locations** and press ENTER.

   The Locations screen displays.

```
change locations                                                        Page 1 of 3
                                     LOCATIONS

                     ARS Prefix 1 Required for 10-Digit NANP Calls? _

 Number   Name                    Timezone   Daylight-Savings         Number Plan
                                  Offset     Rule                     Area Code

    1     ____________________    _ __:__    __                       ___
    2     ____________________    _ __:__    __                       ___
    3     ____________________    _ __:__    __                       ___
    4     ____________________    _ __:__    __                       ___
    5     ____________________    _ __:__    __                       ___
    6     ____________________    _ __:__    __                       ___
    7     ____________________    _ __:__    __                       ___
    8     ____________________    _ __:__    __                       ___
    9     ____________________    _ __:__    __                       ___
   10     ____________________    _ __:__    __                       ___
   11     Germany_____________    + _8:__    __                       ___
   12     ____________________    _ __:__    __                       ___
   13     ____________________    _ __:__    __                       ___
   14     ____________________    _ __:__    __                       ___
```

2. On line 11, in the Name field, enter switch associated with the location (for instance, **Germany**).
3. In the Timezone Offset field, enter **+** to indicate the time is ahead of the system time.
4. In the Timezone Offset field, enter **08** for the number of hours difference between this location and system time.
5. In the Daylight Savings field, enter **1** if this country has daylight savings.
6. Press ENTER to save your changes.

7. Type **change ip-network-map** and press ENTER.

   The IP Address Mapping screen displays.

```
change ip-network-map                                              1 of X

                            IP ADDRESS MAPPING

                                                Subnet
 From IP Address    (To IP Address   or Mask)   Region
    .   .   .          .   .   .
```

8. In the From IP Address field, enter the IP address for the remote station in Germany.
9. In the To IP Address field, enter the IP address of the remote system.
10. In the Subnet or Mask field, enter the subnet mask value of your network.
11. In the Region field, enter a number that is not being used. In this example, enter **100**.
12. Press ENTER to save your changes.
13. Type **ip-network-region-100** and press ENTER.

    The IP Network Region screen displays.

```
change ip-network-region 100                                  Page   1 of   2
                                IP Network Region

      Region: 1
        Name:

Audio Parameters                 Direct IP-IP Audio Connections? n
   Codec Set: 1                            IP Audio Hairpinning? y
    Location: 3
UDP Port Range                                     RTCP Enabled? y
         Min: 2048               RTCP Monitor Server Parameters
         Max: 65535               Use Default Server Parameters? n
                                              Server IP Address: ___.___.___.___
DiffServ/TOS Parameters                             Server Port: _____
 Call Control PHB Value:               RTCP Report Period(secs): __
   VoIP Media PHB Value:
          BBE PHB Value:          Resource Reservation Parameters
                                                      RSVP Enabled? y
             802.1p/Q Enabled? n            RSVP Refresh Rate(secs):
 Call Control 802.1p Priority:      Retry upon RSVP Failure Enabled? y
   VoIP Media 802.1p Priority:                  RSVP Profile: guaranteed-service
                  802.1Q VLAN:
```

14. In the Name field, enter the location name for familiarity.
15. In the Location field, enter the number from the Locations screen. In this example, it was **11**.
16. Press Next Page.

    Page 2 (Inter Network Region Connection Management) displays.

```
change ip-network-region 100                                   Page 2 of 2

                       Inter Network Region Connection Management

Region                                (Group of 32)
        1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2
001-032 1
033-064
065-096
097-128                    1
129-160
161-192
193-224
```

17. Notice in the Region column 001-032, that a “1” displays under column 1 indicating that Network Region 100 (Germany) is connected to Network Region 1 (New York) using Codec Set 1.
18. In row 001-032, under column 1, enter 1 for New York.
19. Press ENTER to save your changes.

2420 DCP Telephone

Downloading Firmware to a 2420 DCP Telephone

You can copy updated application code into ACP via TFTP over a TCP/IP connection. This eliminates the need to physically remove the telephone and send it to the factory for the firmware update. This feature is available on all Avaya MultiVantage platforms.

Before you start

1. Type **change node-name ip** and press ENTER.

   The IP Node Names screen appears.
2. Administer the TFTP server node name and the local node name (CLAN) and IP address.
3. Press ENTER to save your changes.

4. Type **change ip-interfaces** and press ENTER.

   The IP Interfaces screen appears.

5. Administer the CLAN ethernet interface or processor CLAN.
6. Press ENTER to save your changes.

File Download to the switch

1. File is placed on TFTP server using TFTP, FTP, HTTP or another file transfer program.
2. Log into SAT and enter **change tftp-server** and press ENTER.

   The TFTP Server Configuration screen appears.

```
change tftp-server                                          Page 1 of 1
                        TFTP Server Configuration

      Local Node Name:
TFTP Server Node Name:
     TFTP Server Port: 69
     File to Retrieve:

          File Status:
            File Size:
   Filename in Memory:
```

3. In the Local Node Name field, enter the valid local node name from the IP Node Names screen.

   The node must be assigned to a CLAN ip-interface or **procr** (processor CLAN).

4. In the TFTP Server Node Name field, enter the valid TFTP server node name from the IP Nodes Names screen.
5. In the TFTP Server Port field, enter the TFTP server port number from where the file download begins.
6. In the File to Retrieve field, enter the name of the file to be retrieved.
7. Press ENTER to save your changes.

   The file transfer begins.

8. Enter **display tftp-server** and press ENTER to view the status of the file transfer.

   "File download successful" message appears when the file transfer completes. It also displays the file size and the file name in memory.

File Download to the 2420 DCP Telephone

Before you start

You must have console permissions to download someone else's phones.

NOTE:
Steps 1 through 3 need be done only once to set up the FAC for file downloads. Thereafter, start at step 4 to download files.

1. Type change feature-access-codes and press ENTER.
2. In the Station Firmware Download Access Code field, enter *36.

   This number can be any valid FAC as defined in the dial plan.
3. Press ENTER to save your changes.
4. Take the 2420 DCP phone off-hook.
5. Dial the Station Firmware Download FAC (for instance, *36).
6. Press # if you are dialing from the target station (or dial the phone's extension to be downloaded).
7. Place the phone on-hook within 4 seconds after the confirmation tone.

   The phone is placed in a busy-out state (not able to make or receive calls) and displays "Firmware Download in Progress", the amount of the file downloaded, and a timer. The phone displays error messages and a success message before rebooting.
8. When the download completes, the phone reboots and is released from the busy-out state.

Managing phone features

4

This section provides generic instructions for adding any feature button. Because you may need more information to decide which feature buttons you want to assign to a user or group of users, we included the Telephone feature buttons table in this section. This table lists all of the feature buttons that are available on the Avaya MultiVantage solution.

After the list of feature buttons, we included several procedures that explain how to set up the specific feature buttons that require special treatment or considerations.

Adding feature buttons

Once you add a phone to the system, you can use the station screen to change the settings for the phone, such as adding or changing feature button assignments. The system allows you to assign features or functionality to each programmable button. It is up to you to decide which features you want for each phone and which feature you want to assign to each button.

NOTE:
If you have 6400-series phones, your users can administer some of their own feature buttons. See ‘‘Setting up Terminal Self Administration’’ on page 158 for more information.

Instructions

To assign feature buttons:

1. Type **change station *nnnn*** and press ENTER, where *nnnn* is the extension for the phone you want to modify.

   The Station screen appears.

2. Press NEXT PAGE until you locate the Feature Button Assignment fields.

   Some phones have several feature button groups. Make sure that you are changing the correct button. If you do not know which button on the phone maps to which button-assignment field, refer to your phone's manual, or refer to ''Phone reference'' on page 1273.

3. Move the cursor to the field you want to change.
4. Type the button name that corresponds to the feature you want to add.

   To determine feature button names, press HELP or refer to ''Telephone feature buttons'' on page 129.

5. Press ENTER to save your changes.

Some phones have default assignments for buttons. For example, the following figure shows that the 8411D includes defaults for 12 softkey buttons. It already has assignments for features like Leave Word Calling and Call Forwarding.

If you do not use an alias, you can easily assign different features to these buttons if you have different needs.

If you use an alias you must leave the default softkey button assignments. The system allows you to change the button assignments on the screen and the features work on the alias phone, however the labels on the display do not change.

```
                                     STATION
SOFTKEY BUTTON ASSIGNMENTS

 1: lwc-store
 2: lwc-cancel
 3: auto-cback
 4: timer
 5: call-fwd    Ext: ________
 6: call-park
 7: date-time
 8: priority
 9: abr-prog
10: abr-spchar  Char: ~p
11: abr-spchar  Char: ~m
12: abr-spchar  Char: ~w
```

Telephone feature buttons

The following table provides descriptions of the feature buttons that you can administer on multiappearance telephones. It also lists the administrable software names and recommended button label names. Display buttons support telephones equipped with alphanumeric displays. Note that some buttons may require 1-lamp or 2-lamp buttons. Some buttons are not allowed on some systems and on some phones.

Table 2. Telephone feature buttons

Button name	Button label	Description	Maximum
abr-prog	AbrvDial Program	Abbreviated Dialing Program: allows users to program abbreviated dialing and autodial buttons or to store or change numbers in a personal list or group list associated with the station.	1 per station
abr-spchar	AbrvDial (char)	Abbreviated Dialing Special Character: allows users to enter an associated special character [~, ~m (mark), ~p (pause), ~s (suppress), ~w (wait for dial tone), or ~W (wait forever)] when programming an abbreviated dialing list entry.	1 each per station
abrdg-appr (Ext: _____)	(extension)	Bridged Appearance of an analog phone: allows the user to have an appearance of a single-line telephone extension. Assign to a 2-lamp appearance button.	Depends on station type
abrv-dial (List: __ DC: __)	AD	Abbreviated Dialing: dials the stored number on the specified abbreviated dialing list. List: specify the list number 1 to 3 where the destination number is stored DC: specify the dial code for the destination number	1 per AD list per dial code
abrv-ring	AR	Abbreviated and Delayed Ringing: allows the user to trigger an abbreviated or delayed transition for calls alerting at an extension.	

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
ac-alarm	AC Alarm	Administered Connection alarm notification: allows the user to monitor when the number of failures for an administered connection has met the specified threshold.	1 per station
aca-halt	Auto-Ckt Assure	Automatic Circuit Assurance (*display button*): allows users of display telephones to identify trunk malfunctions. The system automatically initiates a referral call to the telephone when a possible failure occurs. When the user presses ACA Halt, the system turns off ACA monitoring for the entire system. The user must press ACA Halt again to restart monitoring.	1 per system
account	Acct	Account: allows users to enter Call Detail Recording (CDR) account codes. CDR account codes allow the system to associate and track calls according to a particular project or account number.	
admin	Admin	Administration: allows a user to program the feature buttons on their 6400-series telephone.	
after-call Grp:___	After Call Work	After Call Work Mode: allows an agent to temporarily be removed from call distribution in order for the agent to finish ACD-related activities such as completing paperwork. Grp: specify the ACD split group number.	1 per split group
alrt-agchg	Alert Agent	Alert Agent: indicates to the agent that their split/skill hunt group changed while active on a call. This button blinks to notify the agent of the change.	1 per station

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
alt-frl	Alt FRL	Alternate Facility Restriction Level (FRL): activates or deactivates an alternate facility restriction level for the extension.	1 per system
ani-requst	ANI Request	Automatic Number Identification Request: allows the user to display the calling party’s number from incoming trunks during the voice state of call. The trunk must support this functionality.	1 per station
assist (Group: __)	Assist	Supervisory Assistance: used by an ACD agent to place a call to a split supervisor. Group: specify the ACD split group number.	1 per split group
asvn-halt	asvn-halt	Authorization Code Security Violation Notification: activates or deactivates call referral when an authorization code security violation is detected.	1 per system
atd-qcalls	AQC	Attendant Queue Calls (display button): tracks the number of calls in the attendant group’s queue and displays the queue status. Assign this button to any user who you want to backup the attendant.	1 per station
atd-qtime	AQT	Attendant Queue Time (display button): tracks the calls in the attendant group’s queue according to the oldest time a call has been queued, and obtains a display of the queue status.	1 per station
aut-msg-wt (Ext: ___)	Message (name or ext #)	Automatic Message Waiting: associated status lamp automatically lights when an LWC message has been stored in the system for the associated extension (can be a VDN). This lamp will not light on the mapped-to physical station for messages left for virtual extensions.	1 per aut-mst-ext

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
auto-cback	Auto CallBack	Automatic Call Back: when activated, allows inside user who placed a call to a busy or unanswered telephone to be called back automatically when the called telephone becomes available to receive a call.	1 per station
auto-icom (Group: __)	Auto (name or ext #)	Automatic Intercom: places a call to the station associated with the button. The called user receives a unique alerting signal, and a status lamp associated with a Intercom button flashes. Grp: Intercom — Auto-Icom group number. This extension and destination extension must be in the same group.	1 per group per dial code
auto-in (Group: __)	Auto In	Auto-In Mode: allows the user to become automatically available for new ACD calls upon completion of an ACD call. Grp: The split group number for ACD.	1 per split group
auto-wkup	Auto Wakeup	Automatic Wakeup (*display button*): allows attendants, front-desk users, and guests to request a wakeup call to be placed automatically to a certain extension (may not be a VDN extension) at a later time.	1 per station
autodial	Autodial	Allows a user to dial a number that is not part of a stored list.	
aux-work (Group: __)	Auxiliary Work	Auxiliary Work Mode: removes agent from ACD call distribution in order to complete non-ACD-related activities. Grp: The split group number for ACD.	1 per split group

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
brdg-appr (Btn: __ Ext: ___)	(extension)	Bridged Call Appearance: provides an appearance of another user's extension on this telephone. For example, an assistant might have a bridged appearance of their supervisor's extension. The bridged appearance button functions exactly like the original call appearance, for instance it indicates when the appearance is active or ringing. You can assign brdg-appr buttons only to 2-lamp appearance buttons. You must indicate which extension and which call appearance button the user wants to monitor at this phone.	Depends on station type
btn-view	Button View	Button View: allows users to view, on the phone's display, the contents of any feature button. Button View does more than the "View" or "stored-num" feature button; these only display what is contained in abbreviated dialing and autodial buttons. When the user presses the btn-view button and then a specific feature button, they see the feature name and any auxiliary data for that button. This allows users to review the programming of their feature buttons. You can assign this soft-key button to any 6400-, 7400-, or 8400-series display telephone.	

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
busy-ind (TAC/Ext: __)	Busy	Busy Indication: indicates the busy or idle status of an extension, trunk group, terminating extension group (TEG), hunt group, or loudspeaker paging zone. Users can press the busy-ind button to dial the specified extension. You can assign this button to any lamp button and must specify which Trunk or extension the user wants to monitor.	1 per TAC/Ext
call-appr	*extension*	Call Appearance: originates or receives calls. Assign to a 2-lamp appearance button.	Depends on station type
call-disp	Return Call	Call Displayed Number (*display button*): initiates a call to the currently displayed number. The number may be from a leave word calling message or a number the user retrieved from the Directory.	1 per station
call-fwd (Ext: ___)	Call Forwarding	Activates or deactivates Call Forwarding All Calls.	
call-park	Call Park	Allows the user to place the current call in the call park state so it can be retrieved from another phone.	1 per station
call-pkup	Call Pickup	Allows the user to answer a call that is ringing in the user's pickup group.	1 per station
call-timer	CTime	Used only on the 6400 sets. Allows users to view the duration of the call associated with the active call appearance button.	1 per station
callr-info	Caller Info	(*display button*) Used with Call Prompting to allow users to display information collected from the originator.	1 per station

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
cas-backup	CAS Backup	Centralized Attendant Service Backup: used to redirect all CAS calls to a backup extension in the local branch if all RLTs are out-of-service or maintenance busy. The associated status lamp indicates if CAS is in the backup mode.	1 per station
cdr1-alrm	CDR 1 Failure	CDR Alarm: associated status lamp is used to indicate that a failure in the interface to the primary CDR output device has occurred.	1 per station
cdr2-alrm	CDR 2 Failure	CDR Alarm: associated status lamp is used to indicate that a failure in the interface to the secondary CDR output device has occurred.	1 per station
cfwd-bsyda	Call Forwarding bsyda (Ext)	Call Forward Busy/Don't Answer: activates and deactivates call forwarding for calls when the extension is busy or the user does not answer.	
check-in	Check In	Check In (*display button*): changes the state of the associated guest room to occupied and turns off the outward calling restriction for the guest room's station.	1 per station
check-out	Check Out	Check Out (*display button*): Changes the state of the associated guest room to vacant and turns on the outward calling restriction for the guest room's station. Also clears (removes) any wake-up request for the station.	1 per station
clk-overid	Clocked Override	Clocked Manual Override (*display button*): used in association with Time of Day Routing to override the routing plan in effect for the activating user. The routing plan is overridden for a specified period of time.	1 per station

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
conf-dsp	Conference Display	Allows a user to display information about each party of a conference call. This button can be assigned to stations and attendant consoles.	1 per station
consult	Consult	The Consult button allows a covering user, after answering a coverage call, to call the principal (called party) for private consultation. Activating Consult places the caller on hold and establishes a private connection between the principal and the covering user. The covering user can then add the caller to the conversation, transfer the call to the principal, or return to the caller.	1 per station
cov-cback	Coverage Callback	Allows a covering party to store a leave word calling message for the principal (called party).	1 per station
cov-msg-rt	Covr Msg Retrieve	Coverage Message Retrieval (*display button*): places a covering station into the message retrieval mode for the purposes of retrieving messages for the group.	1 per station
cpn-blk	CPN Block	Blocks the sending of the calling party number for a call.	1 per station
cpn-unblk	CPN Unblock	Deactivates calling party number (CPN) blocking and allows the CPN to be sent for a single call.	1 per station

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
crss-alert	Crisis	Crisis Alert (*display button*): provide this button to the telephones or consoles that you want to notify when any user makes an emergency call. (You define which calls are emergency calls on the AAR/ARS Analysis screen by setting the Call Type to **alrt**.) After a user receives an alert, they can press the crss-alert button to disable the current alert. If tenant partitioning is active, the attendants within a partition can receive emergency notification only from callers in the same partition.	1 per station 10 per system
data-ext	Data (data ext #)	Data Extension: sets up a data call. May be used to pre-indicate a data call or to disconnect a data call. May not be a VDN or ISDN-BRI extension.	1 per data-extension group
date-time	Date Time	Date and Time (*display button*): displays the current date and time. Do not assign this button to 6400-series display phones as they normally show the date and time.	1 per station
delete-msg	Delete Message	Delete message (*display button*): deletes a stored message that is currently on the display.	1 per station
dial-icom (Grp: ___)	Dial Icom	Dial Intercom: accesses the intercom group assigned to the button. Grp: Intercom — Dial (Dial Icom) group number.	1 per group
did-remove	DID Remove	DID Remove (*display button*): allows DID assignments to be removed.	1 per station
did-view	DID View	DID View (*display button*): allows DID assignments to be displayed and changed. Allows choice between XDID and XDIDVIP numbers.	1 per station

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
directory	Directory	Directory (*display button*): allows users with display telephones to access the system directory, use the touch-tone buttons to key in a name, and retrieve an extension from the directory. The directory contains the names and extensions that you have assigned to the telephones administered in your system. If you assign a directory button, you should also assign a Next and Call-Disp button to the phone. These buttons allow the user to navigate within the directory and call an extension once they find the correct one. Note that Vector Directory Numbers do not appear in the Directory.	1 per station
dir-pkup	dir-pkup	Directed call pickup: allows the user to answer a call ringing at another extension without having to be a member of a pickup group.	
disp-chrg	Display Charge	Provides your display phone with a visual display of accumulated charges on your current telephone call. Used exclusively outside the U.S. and Canada.	1 per station
disp-norm	Local/ Normal	Normal (*display button*): Toggles between LOCAL display mode (displays time and date) and NORMAL mode (displays call-related data). LED off = LOCAL mode and LED on = NORMAL.	1 per station
dn-dst	Do Not Disturb	Places the user in the do not disturb mode.	1 per station
drop	Drop	Allows users to drop calls. Users can drop calls from automatic hold or drop the last party they added to a conference call.	

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
exclusion	Exclusion	Exclusion: allows multiappearance telephone users to keep other users with appearances of the same extension from bridging onto an existing call. If the user press the EXCLUSION button while other users are already bridged onto the call, the other users are dropped. There are two means of activating exclusion. ■ Manual Exclusion — when the user presses the EXCLUSION button (either before dialing or during the call). ■ Automatic Exclusion — as soon as the user picks up the handset. To turn off Automatic Exclusion during a call, the user presses the EXCLUSION button. To use Automatic Exclusion, set the Automatic Exclusion by COS field to **y** on the Feature-Related System Parameters screen.	1 per station
ext-dn-dst	Do Not Disturb Ext	Extension — Do Not Disturb (*display button*): used by the attendant console or hotel front desk display phone to activate do not disturb and assign a corresponding deactivate time to an extension.	1 per station
fe-mute	Far End Mute for Conf	Allows a user to mute a selected party on a conference call. This button can be assigned to stations and attendant consoles.	1 per station

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
flash	Flash	1) Allows a station on a trunk call with Trunk Flash to send a Trunk Flash signal to the far end (e.g., Central Office); 2) allows a station on a CAS main call to send a Trunk Flash signal over the connected RLT trunk back to the branch to conference or transfer the call.	1 per station
goto-cover	Go To Cover	Go To Coverage: sends a call directly to coverage instead of waiting for the called inside-user to answer. Go to Cover forces intercom and priority calls to follow a coverage path. **NOTE:** Go to Cover cannot be activated for calls placed to a Vector Directory Number extension. Go to Cover can be used to force a call to cover to a VDN if the called principal has a VDN as a coverage point.	1 per station
grp-dn-dst	Do Not Disturb Grp	Group Do Not Disturb (*display button*): places a group of phones into the do not disturb mode.	1 per station
grp-page (Number:___)	GrpPg	Allows users to make announcements to groups of stations by automatically turning on their speakerphones. Number: The extension of the page group.	
headset	Headset	Signals onhook/offhook state changes to the switch. The green LED is on for offhook state and off (dark) for onhook state.	
hunt-ns (Grp: ___)	Hunt Group	Hunt-Group Night Service: places a hunt-group into night service. Grp: Hunt group number.	3 per hunt group

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
in-call-id (Type: __ Grp: ___)	Coverage (group #, type, name, or ext #)	The Coverage Incoming Call Identification (ICI) button allows a member of a coverage answer group or hunt group to identify an incoming call to that group even though the member does not have a display telephone. In the Type field, enter **c** for coverage answer groups and type of **h** for a hunt group. In the Grp field, enter the group number.	1 per group-type per group
inspect	Inspect Mode	Inspect (*display button*): allows users on an active call to display the identification of an incoming call. Inspect also allows users to determine the identification of calls they placed on Hold.	1 per station
int-aut-an	IntAutoAns	Internal Automatic Answer: causes any hybrid or digital station to automatically answer incoming internal calls.	1 per station
last-numb	LastNumb Dialed	Last Number Dialed (redial): originates a call to the number last dialed by the station.	1 per station
lic-error	License Error	License-Error: indicates a major License File alarm. Pressing the button does not make the light go out. The button goes out only after the error is cleared and the switch returns to License-Normal Mode. You can administer this button on phones and attendant consoles.	1 per phone 20 per system (Server CSI/ Server SI) 30 per system (Server R)

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
link-alarm (link# ___)	Link Failure (link #)	Link Alarm: associated status lamp indicates that a failure has occurred on one of the Processor Interface circuit pack data links. Link: Link number — 1 to 8 for multi-carrier cabinets or 1 to 4 for single-carrier cabinets.	8 per station
lsvn-halt	Login SVN	Login Security Violation Notification: activates or deactivates referral call when a login security violation is detected.	1 per system
lwc-cancel	Cancel LWC	Leave Word Calling Cancel: cancels the last leave word calling message originated by the user.	1 per station
lwc-lock	Lock LWC	Leave Word Calling Lock: locks the message retrieval capability of the display module on the station.	1 per station
lwc-store	LWC	Leave Word Calling Store: leaves a message for the user associated with the last number dialed to return the call to the originator.	1 per station
major-alrm	Major Hdwe Failure	Major Alarm: assign to a status lamp to notify the user when major alarms occur. Major alarms usually require immediate attention.	1 per station
man-msg-wt (Ext: ___)	Msg Wait (name or ext #)	Manual Message Waiting: allows a multiappearance telephone user to press a button on their telephone in order to light the Manual Message Waiting button at another telephone. You can administer this feature only to pairs of telephones, such as an assistant and an executive. For example, an assistant can press the man-msg-wt button to signal the executive that they have a call.	None

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
man-overid (TOD: _)	Immediate Override	Immediate Manual Override (*display button*): allows the user (on a system with Time of Day Routing) to temporarily override the routing plan and use the specified TOD routing plan. TOD: specify the routing plan the user wants to follow in override situations.	1 per station
manual-in (Group: __)	Manual In	Manual-In Mode: prevents the user from becoming available for new ACD calls upon completion of an ACD call by automatically placing the agent in the after call work mode. Grp: The split group number for ACD.	1 per split group
mct-act	MCT Activation	Malicious Call Trace Activation: sends a message to the MCT control extensions that the user wants to trace a malicious call. MCT activation also starts recording the call, if your system has a MCT voice recorder.	

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
mct-contr	MCT Control	Malicious Call Trace Control: allows the user to take control of a malicious call trace request. Once the user becomes the MCT controller, the system stops notifying other MCT control extensions of the MCT request. **NOTE:** To add an extension to the MCT control group, you must also add the extension on the ''Extensions Administered to have an MCT-Control Button'' screen. When the user presses the MCT Control button, the system first displays the called party information. Pressing the button again displays the rest of the trace information. The MCT controller must dial the MCT Deactivate feature access code to release control.	
mf-da-intl	Directory Assistance	Multifrequency Operator International: allows users to call Directory Assistance.	1 per station
mf-op-intl	CO attendant	Multifrequency Operator International: allows users to make international calls to the CO attendant.	1 per station
mj/mn-alrm	Maj/Minor Hdwe Failure	Minor Alarm: assign to a status lamp to notify the user when minor or major alarms occur. Minor alarms usually indicate that only a few trunks or a few stations are affected.	1 per station
mm-basic	MM Basic	Multimedia Basic: used to place a multimedia complex into the “Basic” mode or to return it to the “Enhanced” mode.	1 per station

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
mm-call	MM Call	Multimedia Call: used to indicate a call is to be a multimedia call.	1 per station
mm-cfwd	MM CallFwd	Multimedia Call Forward: used to activate forwarding of multimedia calls as multimedia calls, not as voice calls.	1 per station
mm-datacnf	MM Datacnf	Multimedia Data Conference: used to initiate a data collaboration session between multimedia endpoints; requires a button with a lamp.	1 per station
mm-multnbr	MM MultNbr	Indicate that the user wants to place calls to 2 different addresses using the 2 B-channels.	1 per station
mm-pcaudio	MM PCAudio	Switches the audio path from the telephone (handset or speakerphone) to the PC (headset or speakers/microphone).	1 per station
msg-retr	Message Retrieve	Message Retrieval (*display button*): places the station's display into the message retrieval mode.	1 per station
mwn-act	Message Waiting Act.	Message Waiting Activation: lights a message waiting lamp on an associated station.	1 per station
mwn-deact	Message Waiting Deact	Message Waiting Deactivation: dims a message waiting lamp on an associated station.	1 per station
next	Next	Next (*display button*): steps to the next message when the phone's display is in Message Retrieval or Coverage Message Retrieval mode. Shows the next name when the phone's display is in the Directory mode.	1 per station
night-serv	Night Serv	Night Service Activation: toggles the system in or out of Night Service mode.	1 per station

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
noans-alrt	RONA	Redirection on No Answer Alert: indicates a Redirection on No Answer timeout has occurred for the split.	1 per split
no-hld-cnf	No Hold Conference	No Hold Conference: can be assigned to all digital station types but not to an attendant console.	1 per station
normal	Normal Mode	Normal (*display button*): places the station's display into normal call identification mode.	1 per station
off-bd-alm	Off board alarm	Off board Alarm: associated status lamp lights if an off-circuit pack major, minor, or warning alarm is active on a circuit pack. Off-board alarms (loss of signal, slips, misframes) relate to problems on the facility side of the DS1, ATM, or other interface.	
per-COline (Grp: ___)	CO Line (line #)	Personal CO Line: allows the user to receive calls directly via a specific trunk. Grp: CO line group number.	1 per group
pms-alarm	PMS Failure	Property Management System alarm: associated status lamp indicates that a failure in the PMS link occurred. A major or minor alarm condition raises the alarm.	1 per station
pr-awu-alm	Auto Wakeup Alm	Automatic Wakeup Printer Alarm: associated status lamp indicates that an automatic wakeup printer interface failure occurred.	1 per station
pr-pms-alm	PMS Ptr Alarm	PMS Printer Alarm: associated status lamp indicates that a PMS printer interface failure occurred.	1 per station
pr-sys-alm	Sys Ptr Alarm	System Printer Alarm: associated status lamp indicates that a system printer failure occurred.	1 per station

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
print-msgs	Print Msgs	Print Messages: allows users to print messages for any extension by pressing the button and entering the extension and a security code.	1 per station
priority	Priority Call	Priority Calling: allows a user to place priority calls or change an existing call to a priority call.	1 per station
q-calls (Grp: ___)	NQC	Queue Calls: associated status lamp flashes if a call warning threshold has been reached. Grp: Group number of hunt group.	1 per hunt group per station
q-time (Grp: ___)	OQT	Queue Time: associated status lamp flashes if a time warning threshold has been reached. Grp: Group number of hunt group.	1 per hunt group per station
release	Release	Releases an agent from an ACD call.	1 per station
ringer-off	Ringer Cutoff	Ringer-Cutoff: silences the alerting ringer on the station.	1 per station
rs-alert	System Reset Alert	The associated status lamp lights if a problem escalates beyond a warm start.	1 per station
rsvn-halt	rsvn-halt	Remote Access Barrier Code Security Violation Notification Call: activates or deactivates call referral when a remote access barrier code security violation is detected.	1 per system
scroll	Scroll	Scroll (*display button*): allows the user to select one of the two lines (alternates with each press) of the 16-character LCD display. Only one line displays at a time.	1 per station

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
send-calls (Ext: ___)	Send All Calls	Send All Calls allows users to temporarily direct all incoming calls to coverage regardless of the assigned call-coverage redirection criteria. Assign to a lamp button.	
send-term	Send All Calls-TEG	Send All Calls For Terminating Extension Group: allows the user to forward all calls directed to a terminating extension group.	1 per TEG
serv-obsrv	Service Observing	Service Observing: activates Service Observing. Used to toggle between a listen-only and a listen-talk mode.	1 per station
signal (Ext: ___)	Signal (name or ext #)	Signal: allows the user to use one button to manually signal the associated extension. The extension cannot be a VDN extension.	1 per signal extension
ssvn-halt	ssvn-halt	Toggle whether or not station security code violation referrals are made to the referral destination.	1 per station
sta-lock	Station Lock	Station Lock: when enabled, no outgoing calls can be made from the phone.	1 per station
stored-num	Stored Number	(*display button*) Places the station's display into the stored number mode.	1 per station
stroke-cnt	ACD SD Stroke Count	Automatic Call Distribution Single Digit Stroke Count: sends a message to CMS to increment a stroke count number.	1 per station
term-x-gr (Grp: ___)	Term Grp (name or ext #)	Terminating Extension Group: provides one or more extensions. Calls may be received but not originated with this button. Grp: TEG number.	1 per TEG
timer	Timer	(*display button*) Starts a clock on the station to display elapsed time.	1 per station
togle-swap	Conf/Trans Toggle-Swap	Allows a user to toggle between two parties before completing a conference or a transfer.	1 per station

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
trk-ac-alm	FTC Alarm	Facility Test Call Alarm: associated status lamp lights when a successful Facility Test Call (FTC) occurs.	
trk-id	Trunk ID	Trunk Identification (*display button*): identifies the tac (trunk access code) and trunk member number associated with a call.	1 per station
trunk-name	Trunk Name	(*display button*) Displays the name of the trunk as administered on the CAS Main or on a switch without CAS.	1 per station
trunk-ns (Grp: ___)	Trunk Grp	Trunk-Group Night Service: places a trunk-group into night service. Grp: Trunk group number.	1 per trunk group
verify	Verify	Busy Verification: allows users to make test calls and verify a station or a trunk.	1 per station
vip-chkin	VIP Check-in	VIP Check-in (*display button*): allows user to assign the XDIDVIP number to the room extension.	1 per station
vip-retry	VIP Retry	VIP Retry: starts to flash when the user places a VIP wakeup call and continues to flash until the call is answered. If the VIP wakeup call is not answered, the user can press the VIP Retry button to drop the call and reschedule the VIP wakeup call as a classic wakeup call. To assign this button, you must have both Hospitality and VIP Wakeup enabled.	1 per station
vip-wakeup	VIP Wakeup	VIP Wakeup: flashes when a VIP wakeup reminder call is generated. The user presses the button to place a priority (VIP) wakeup call to a guest. To assign this button, you must have both Hospitality and VIP Wakeup enabled.	1 per station

Continued on next page

Table 2. Telephone feature buttons (Continued)

Button name	Button label	Description	Maximum
voa-repeat	VOA repeat	VDN of Origin Announcement. VDN of Origin Announcement must be enabled.	1 per station
vu-display (format: __ ID: __)	VuStats #	VuStats Display: allows the agent to specify a display format for the statistics. If you assign a different VuStats display format to each button, the agent can use the buttons to access different statistics. You can assign this button only to display phones. format: specify the number of the format you want the button to display ID (optional): specify a split number, trunk group number, agent extension, or VDN extension	limited to the number of feature buttons on the phone
whisp-act	Whisper Page Activation	Whisper Page Activation: allows a user to make and receive whisper pages. A whisper page is an announcement sent to another extension who is active on a call where only the person on the extension hears the announcement; any other parties on the call cannot hear the announcement. The user must have a class of restriction (COR) that allows intraswitch calling to use whisper paging.	
whisp-anbk	Answerback	Whisper Page Answerback: allows a user who received a whisper page to respond to the user who sent the page.	
whisp-off	Whisper Page Off	Deactivate Whisper Paging: blocks other users from sending whisper pages to this phone.	
work-code	Work Code	Call Work Code: allows an ACD agent after pressing “work-code” to send up to 16 digits (using the dial pad) to CMS.	1 per station

Adding abbreviated dialing lists

Abbreviated dialing is sometimes called speed dialing. It allows you to dial a short code in place of an extension or phone number.

When you dial abbreviated-dialing codes or press abbreviated-dialing buttons, you access stored numbers from special lists. These lists can be personal (a list of numbers for an individual phone), group (a department-wide list), system (a system-wide list), or enhanced numbers (allows for a longer list of numbers). The version and type of your system determine which lists are available and how many entries you can have on each list.

NOTE:
You can designate all group-number lists, system-number lists, and enhanced-number lists as “privileged.” Calls automatically dialed from a privileged list are completed without class of restriction (COR) or facility restriction level (FRL) checking. This allows access to selected numbers that some phone users might otherwise be restricted from manually dialing. For example, a user may be restricted from making long-distance calls. However, you can program the number of a branch office that is long distance into an AD list as privileged. Then, the user can call this office location using AD, while still being restricted from making other long-distance calls.

SECURITY ALERT:
Privileged group-number, system-number, and enhanced-number lists provide access to numbers that typically would be restricted.

Instructions

As an example, let’s program a new group list:

1. Type **add abbreviated-dialing group next** and press RETURN.

   The Abbreviated Dialing List screen appears. In our example, the next available group list is group 3.

```
                    ABBREVIATED DIALING LIST

                        Group List: 3
    Size (multiple of 5): ___  Program Ext: ________  Privileged? _
DIAL CODE
     01: ________________________
     02: ________________________
     03: ________________________
     04: ________________________
     05: ________________________
```

2. Enter a number (in multiples of 5) in the Size field. This number defines the number of entries on your dialing list.

   For example, if you have 8 phone numbers you want to store in the list, type **10** in the Size field.

3. If you want another user to be able to add numbers to this list, enter their extension in the Program Ext field.

   For example, if you want the user at 4567 to be able to change group list 3, enter **4567** in this field.

4. Enter the phone numbers you want to store, one for each dial code.

   Each phone number can be up to 24 digits long.

5. Press ENTER to save your changes.

You can display your new abbreviated-dialing list to verify that the information is correct or print a copy of the list for your paper records.

Once you define a group list, you need to define which stations can use the list. For example, let's set up station 4567 so it has access to the new group list.

To give station 4567 access to the group list:

1. Type **change station 4567** and press RETURN.

   The Station screen for extension 4567 appears.

2. Press NEXT PAGE to get to the Abbreviated Dialing List fields.

```
                                     STATION
SITE DATA
     Room: _______                                              Headset? n
     Jack: _______                                              Speaker? n
    Cable: _______                                             Mounting? d
    Floor: _______                                          Cord Length: 0
 Building: _______                                            Set Color: ______

 ABBREVIATED DIALING
     List1: group   3__      List2: _________         List3: _________

 BUTTON ASSIGNMENTS
   1: call-appr
   2: call-appr                            4: __________
   3: call-appr                            5: __________
```

3. Type **group** in any of the List fields and press RETURN.

   A blank list number field appears.

4. Type **3** in the list number field.

   When you assign a group or personal list, you must also specify the personal list number or group list number.

5. Press ENTER to save your changes.

The user at extension 4567 can now use this list by dialing the feature access code for the list and the dial code for the number they want to dial. Alternatively, you can assign an abbreviated dialing button to this station that allows the user press one button to dial a specific stored number on one of their three assigned abbreviated lists.

Fixing problems

Problem	Possible causes	Solutions
A user cannot access a dial list.	■ The specific list may not be assigned to the user's phone.	1. Type **display station *nnnn*** and press RETURN, where *nnnn* is the user's extension. 2. Review the current settings of the List1, List2, and List 3 fields to determine if the list the user wants to access is assigned to their phone.
	■ If the user attempted to use a feature access code to access the list, they may have dialed the incorrect feature access code.	1. Type **display feature-access-codes** and press RETURN. 2. Verify that the user is dialing the appropriate feature access code.
	■ If the user attempted to press a feature button, they may have pressed the incorrect feature button.	1. Type **display station *nnnn*** and press RETURN, where *nnnn* is the user's extension. 2. Review the current feature button assignments to determine if the user was pressing the assigned button.
	■ If the user attempted to press the correct feature button, the button may not be set up correctly.	1. Type **display station *nnnn*** and press RETURN, where *nnnn* is the user's extension. 2. Review the current feature button assignments to see if the list number and dial code are correct.

Problem	Possible causes	Solutions
A user complains that using an abbreviated dial list dials the wrong number.	■ The user could be using the wrong dial code. ■ The dial code could be defined incorrectly.	1. Ask the user what number they dialed or button they pressed to determine which list and dial code they attempted to call. 2. Access the dialing list and verify that the number stored for the specific dial code corresponds to the number the user wanted to dial. (For example to access a group list, type **display abbreviated-dialing group *x*** and press RETURN, where *x* is a group list number.) 3. If the user dialed the wrong code, give them the correct code. 4. If the dial code is wrong, press CANCEL and use the appropriate change command to re-access the abbreviated dialing list. Correct the number and press ENTER.

More information

There are limits to the total number of abbreviated dialing list entries, the number of personal dial lists, and the number of group dial lists that your system can store. Because of these limitations, you should avoid storing the same number in more than one list. Instead, assign commonly dialed numbers to the system list or to a group list. You can determine the abbreviated dialing storage capacity, by referring to the System Capacity screen for the abbreviated dialing values (**display capacity**).

Related topics

For more information, refer to ''Abbreviated Dialing'' on page 1333.

Setting up bridged call appearances

Think of a bridged call appearance as a phone (the primary set) with an extension (the bridged-to appearance). Both phones can be used to call in and out and both show when a line is in use. A call to the primary phone is bridged to a specific appearance, or button, on the secondary phone. The secondary phone retains all its functions, and a specific button is dedicated as the bridged-to appearance from the primary phone.

Bridged call appearances have to be assigned to phones with double-lamp buttons, or lights. The phone types do not need to match, but as much consistency as possible is recommended for all phones in a bridged group. When a call comes in on bridged phones, the buttons assigned to the bridged appearances flash. You can assign as many bridged appearances as there are line appearances on the primary phone, and you can assign ringing (alerting) to one or more of the phones.

Instructions

To create a bridged call appearance:

1. Note the extension of the primary phone.

   A call to this phone lights the button and, if activated, rings at the bridged-to appearance on the secondary phone.

2. If you want to use a new phone for the bridged-to extension, duplicate a station.

   For information refer to ‘‘Using templates to add phones’’ on page 90.

3. Type **change station** and the bridged-to extension and press RETURN.

   The Station screen appears.

```
                                        STATION
FEATURE OPTIONS
          LWC Reception? _______          Auto Select Any Idle Appearance? _
         LWC Activation? _                         Coverage Msg Retrieval? _
            CDR Privacy? _                                    Auto Answer? __
  Redirect Notification? _                               Data Restriction? _
Per Button Ring Control? _                     Idle Appearance Preference? _
  Bridged Call Alerting? _
 Active Station Ringing: ______                 Restrict Last Appearance? _
                                                              Data Module? _

                     XID? _     Fixed TEI? _     TEI: __
            MIM Support? _   Endpt Init? _   SPID: ____     MIM Mtce/Mgt? _
              AUDIX Name: ______
   Messaging Server Name: ______                    Audible Message Waiting? _
                                                          Disp Client Redir? _
                                                  Select Last Used Appearance? _
```

4. Press NEXT PAGE until Per Button Ring Control appears (digital sets only).
   - If you want to assign ringing separately to each bridged appearance, type **y**.
   - If you want all bridged appearances to either ring or not ring, leave the default **n**.
5. Move to Bridge Call Alerting.

   If you want the bridged appearance to ring when a call arrives at the primary phone, type **y**. Otherwise, leave the default **n**.
6. Complete the appropriate field for your phone type.

If. . .	Then. . .
your primary phone is analog	move to the Line Appearance field and enter **abrdg-appr**
your primary phone is digital	move to the Button Assignments field and enter **brdg-appr**

7. Press RETURN.

   Btn and Ext fields appear. If Per Button Ring Control is set to **y** on the digital screen, Btn, Ext, and Ring fields appear.

```
                                        STATION
  SITE DATA
       Room: _______                                                 Headset? n
       Jack: _______                                                 Speaker? n
      Cable: _______                                                Mounting? d
      Floor: _______                                             Cord Length: 0
   Building: _______                                               Set Color: ______

 ABBREVIATED DIALING
     List1: ___________      List2: _________         List3: _________

 HOT LINE DESTINATION

          Abbreviated Dialing List Number (From above 1, 2 or 3):
                                                     Dial Code:
       Line Appearance:  brdg-appr   Btn:     Ext:
```

Screen 3. Station screen (analog)

```
                                    STATION
 SITE DATA
       Room: _______                                             Headset? n
       Jack: _____                                               Speaker? n
      Cable: _____                                              Mounting: d
      Floor: _______                                        Cord Length: 0
   Building: _______                                      Set Color: ______


ABBREVIATED DIALING
   List1: ________         List2: ________         List3: ________

BUTTON ASSIGNMENTS
  1: brdg-appr  Btn:   Ext:      Ring:
  1: brdg-appr  Btn:   Ext:      Ring:
```

Screen 4. Station screen (digital set)

8. Enter the primary phone's button number that you want to assign as the bridged call appearance.

   This button flashes when a call arrives at the primary phone.

9. Enter the primary phone extension.

10. If the Ring field appears:

    - If you want the bridged appearance to ring when a call arrives at the primary phone, type **y**.
    - If you do not want the bridged appearance to ring, leave the default **n**.

11. Press ENTER to save your changes.

To see if an extension has any bridged call appearances assigned, type **list bridge** and the extension, and press RETURN.

More information

Following are a list of example situations where you might want to use bridged appearances.

- A secretary making or answering calls on an executive's primary extension

  These calls can be placed on hold for later retrieval by the executive, or the executive can simply bridge onto the call. In all cases, the executive handles the call as if he or she had placed or answered the call. It is never necessary to transfer the call to the executive.

- Visitor telephones

  An executive may have another telephone in their office that is to be used by visitors. It may be desirable that the visitor be able to bridge onto a call that is active on the executive’s primary extension number. A bridged call appearance makes this possible.

- Service environments

  It may be necessary that several people be able to handle calls to a particular extension number. For example, several users may be required to answer calls to a hot line number in addition to their normal functions. Each user may also be required to bridge onto existing hot line calls. A bridged call appearance provides this capability.

- A user frequently using telephones in different locations

  A user may not spend all of their time in the same place. For this type of user, it is convenient to have their extension number bridged at several different telephones.

Setting up Terminal Self Administration

Terminal Self-Administration (TSA) allows users to administer some of their own feature buttons from their telephones. TSA is available only for 6400-series phones. Users are prompted, via the telephone’s display, to choose features to assign to buttons on their telephones.

Before you start

To prevent users from changing another user's phone administration, you can enable the system-wide option that requires users to enter a station security code before they can administer their phone. To enable this option:

1. Set the Station Security Code for Terminal Self-Administration Required on the Security-Related System Parameters screen to **Y**.

   If you enable this option, the user is prompted for the station security code when they press the Admin button. The user must enter the security code, followed by the pound (#) button or the Done softkey.

Instructions

You need to assign a security code to the user’s Station screen for each user you want to enable access to TSA. You also need to assign the user an Admin feature button.

For example, to assign a security code of 12345678 to extension 4234, complete the following steps:

1. Type **change station 4234** and press RETURN.

   The Station screen for extension 4234 appears.

2. In the Security Code field, type **12345678**.

   You should assign unique security codes for each user. Once you enter the code and move off the field, the system changes the field to '*' for extra security.

3. In one of feature button fields, type **admin**.

   You can assign this button to a feature button or a softkey.

4. Press ENTER to save your changes.

More information

Before a user can enter the TSA Admin mode, their telephone must be completely idle. After a user presses the Admin button and enters a security code (if necessary), they are prompted, via the telephone's display, to choose features to administer to buttons on their telephone.

The user can add, replace, or delete any of the following feature-button types from their telephone.

- CDR Account Code
- Automatic Dial
- Blank
- Call Forwarding
- Call Park
- Call Pickup
- Directed Call Pickup
- Group Page
- Send All Calls
- Activate Whisper Page
- Answerback for Whisper Page
- Whisper Page Off

End-user button changes are recorded to the switch's history log so that remote services can know what translations are changed.

Fixing problems

- When a telephone is in the Admin mode, the telephone cannot accept any calls — the telephone is treated as if it were busy. Also, a user cannot make calls while in the Admin mode.
- Any button state a telephone is in when the telephone enters the Admin mode stays active while the telephone is in the Admin mode.
- ACD agents who wish access to the Admin mode of TSA must be logged off before pressing the Admin button. If they are not logged off when they attempt to enter the Admin mode, they receive a denial (single-beep) tone.
- Call Forwarding can be active and works correctly in the Admin mode. An active Call Forwarding button cannot be removed when the telephone is in the Admin mode.
- Since the telephone must be on-hook to go into the Admin mode, the Headset On/Off button must be in the OFF position.
- A telephone that is in the Admin mode of TSA cannot be remotely unmerged by the PSA feature.

  If a user has Abbreviated and Delayed Ringing active, a call can be silently ringing at a telephone and the user may not realize it. This ringing prevents the user from entering the Admin mode of TSA.

Managing your attendant consoles

5

This section provides an overview to the Avaya attendant consoles. It also explains how to add new consoles, remove consoles, and how to set your system-wide console parameters.

Overview

The attendant console is the main answering position for your organization. The console operator is responsible for answering incoming calls and for efficiently directing or "extending" calls to the appropriate phone.

The attendant console also can allow your attendants to monitor:

- system problems
- toll fraud abuse
- traffic patterns

The number of consoles you can have in your organization varies depending on your MultiVantage system.

302 attendant consoles

MultiVantage supports the following 302 attendant consoles: the 302A/B, 302C, and 302D consoles. You may have a basic or enhanced version of these consoles. Figure 1 on page 162 shows the 302A/B console and Figure 2 on page 163 shows the 302C console. The next two figures show the button layouts on the Feature area and on the optional Selector console.

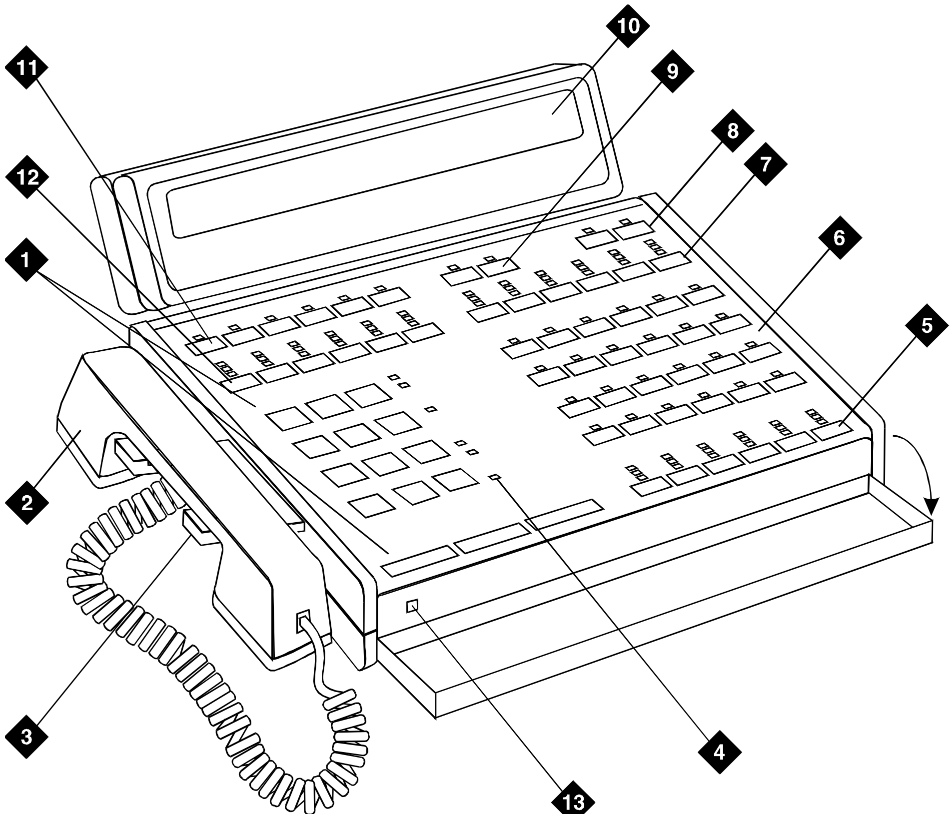


Figure Notes

1. Call processing area
2. Handset
3. Handset cradle
4. Warning lamps and call waiting lamps
5. Call appearance buttons
6. Feature area
7. Trunk group select buttons
8. Volume control buttons
9. Select buttons
10. Console display panel
11. Display buttons
12. Trunk group select buttons
13. Lamp Test Switch

Figure 1. 302A and 302B1 attendant console

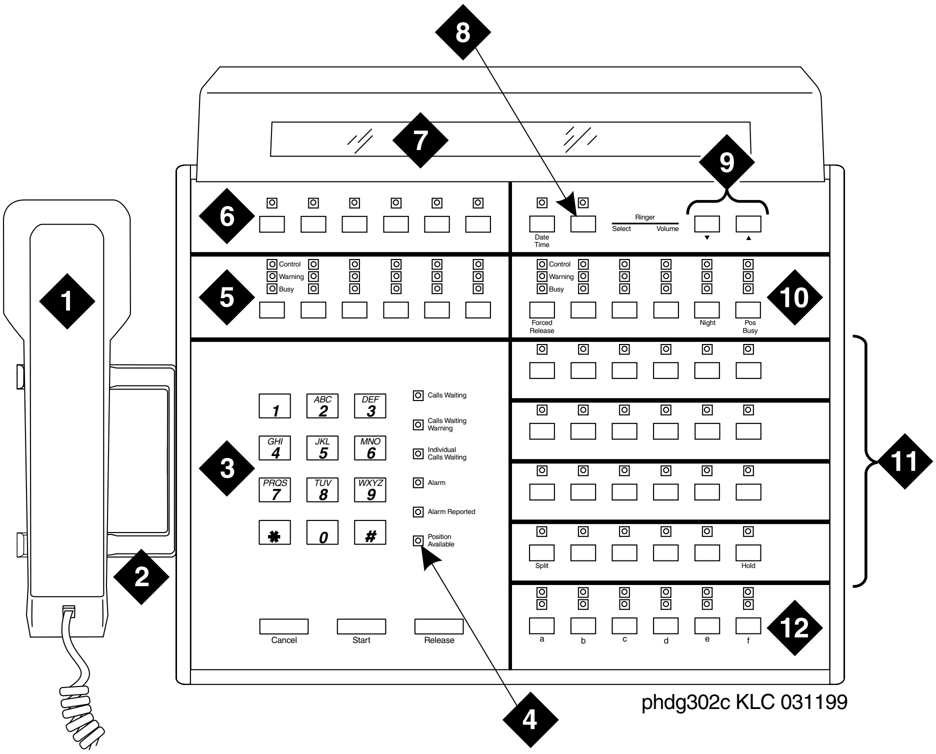


Figure Notes

1. Handset
2. Handset cradle
3. Call processing area
4. Warning lamps and call waiting lamps
5. Outside-line buttons
6. Display buttons
7. Display
8. Select buttons
9. Volume control buttons
10. Outside-line buttons
11. Feature buttons
12. Call appearance buttons

Figure 2. 302C attendant console

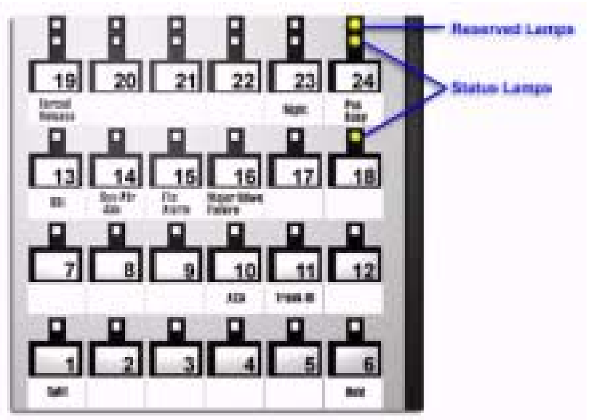


Figure 3. Console feature button layout

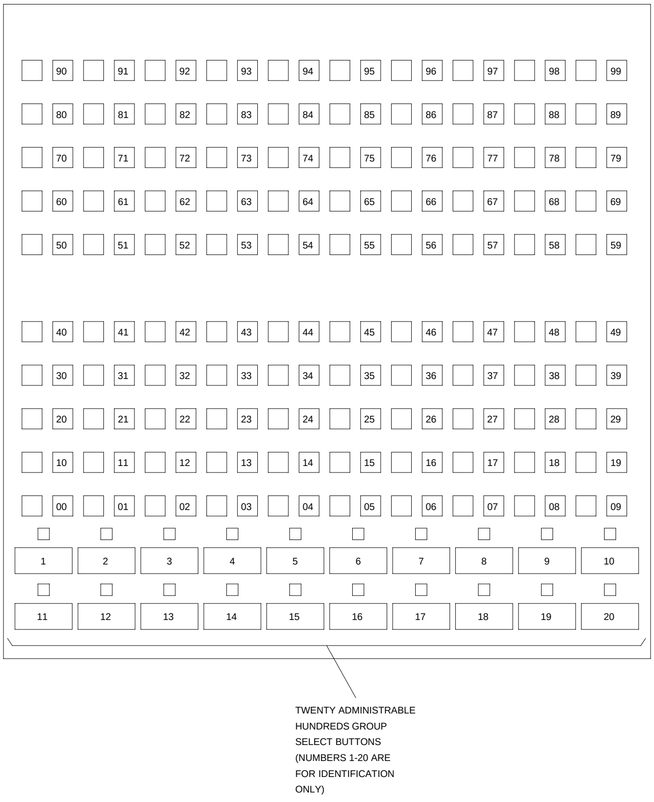


Figure 4. Enhanced Selector Console

302D Console

The 302D console provides the following enhancements to the 302C console:

- Modular handset/headset connection

  The console accepts a standard RJ11, 4-pin modular handset or headset. This connection replaces the quarter-inch, dual-prong handset/headset connection.

- Activate/deactivate push-button

  You can use the push-button on the left side of the console to activate or deactivate the console. A message appears on the console identifying that the button must be pressed to activate the console.

- Two-wire DCP compatibility

  The console is compatible with two-wire DCP circuit packs only, not four-wire DCP circuit packs.

- Headset volume control

  The console can now control the volume of an attached headset.

- Noise expander option

  The console has circuitry to help reduce background noise during pauses in speech from the console end of a conversation. This option is normally enabled.

- Support for Eurofont or Katakana character set

  The console can show the Eurofont or Katakana character set. Administration of these character sets must be coordinated with the characters sent from the switch.

Avaya PC consoles

The Avaya PC Console is a Microsoft Windows-based call handling application for MultiVantage system attendants. It provides an ideal way to increase your productivity and to better serve your customers.

PC Console offers all the call handling capabilities of the hardware-based Avaya 302 attendant console with a DXS module, plus several enhanced features and capabilities. The enhanced features provide you with the ability to see up to six calls at once, and to handle all calls more efficiently.

PC Console also provides a powerful directory feature. You are able to perform searches, display user information, including a photo. You are able to place a call immediately from the directory.

And, because PC Console resides on a Windows-based PC, you are able to use other software applications at the same time. If a call comes in while you are in another application, you are able to handle it immediately.

For more information about the Avaya PC Console, contact your Avaya account team.

SoftConsole IP Attendant

The SoftConsole is a Windows-based application that can replace the 302B hard console. The SoftConsole is similar to PC Console, but it performs call answering and routing through a PC interface via IP. For more information, contact your Avaya account team.

Adding an attendant console

Usually Avaya connects and administers your primary attendant console during cutover. However, you might find a need for a second attendant console, such as a backup console that is used only at night. Note that these instructions do not apply to adding a PC Console or SoftConsole. For more information, refer to the appropriate console documentation.

To add a night-only attendant console, complete the following steps:

1. Type **add attendant 2** and press RETURN.

   The Attendant Console screen appears.

```
                              ATTENDANT CONSOLE 2

        Type: 302           Name: operator
   Extension:              Group: 1            Auto Answer: none
Console Type: night-only      TN: 1            Data Module? n
        Port:                COR: 1       Disp Client Redir? n
                             COS: 1        Display Language: english

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote            Local Remote              Local Remote
 1:                     5:                        9:
 2:                     6:                       10:
 3:                     7:                       11:
 4:                     8:                       12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:            5:            9:           13:           17:
 2:            6:           10:           14:           18:
 3:            7:           11:           15:           19:
 4:            8:           12:           16:           20:
```

2. In the Type field, enter **302**.

   This is the type of attendant console.

3. If you want this attendant to have its own extension, enter one in the Extension field.

   Tip:
   If you assign an extension to the console, the COR and COS that you assign on this Console screen override the COR and COS you assigned on the Console Parameters screen. To avoid unexpected behavior, you should assign the same COR and same COS on both screens.

   If you give your attendants an individual extension, users can call the attendant directly by dialing the extension.

   Individual attendant extensions also allow attendants to use features that an attendant group cannot use — for example, you can assign them to hunt groups.

4. In the Console Type field, enter **night-only**.

   This indicates how this console is used in your organization—as a principal, day only, night only, or day/night console. You can have only one night-time console (night only or day/night) in the system.

5. In the Port field, enter the port address for this console.
6. Type a name to associate with this console in the Name field.
7. In the Direct Trunk Group Select Button Assignments fields, enter trunk access codes for the trunks you want the attendant to be able to select with just one button.
8. If you are using the Enhanced Selector console, assign the Hundreds Select Buttons that you want this console to have.

   If you want this console to be able to access extensions in the range 3500 to 3999, you need to assign them 5 Hundreds Select Buttons: **35** for extensions 3500 to 3599, **36**, **37**, **38**, and **39**.

9. Assign the Feature Buttons that you want the 302 console to have.

   To determine which buttons you can assign to a console, refer to ''Attendant console feature buttons''

   Tip:
   Feature buttons are not numbered top-to-bottom on the attendant console, as you might expect. Button numbers map to physical positions on the console as shown in *Figure 3 on page 164*.

10. Press ENTER to save your changes.

Attendant console feature buttons

The following table lists the feature buttons that you can assign to an attendant console.

Table 3. Attendant console feature buttons

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
Abbreviated Dialing	AD	abrv-dial (List:___ DC:___)	1 per List/DC	1
Administered Connection [status lamp]	AC Alarm	ac-alarm	1	
Alert Agent of Change to Split/Skill Hunt Group	Alert Agent	alrt-agchg	1	
Automatic Call Distribution (ACD)	After Call Work	after-call (Grp. No.__)	N	2
	Assist	assist (Grp. No:__)	1 per split group	2
	Auto In	auto-in (Grp. No.__)	1 per split group	2
	Auxiliary Work	aux-work (Grp. No.__)	1 per split group	2
	Manual-In	manual-in (Grp. No.__)	1 per split group	2
	Release	release	1	
	Work Code	work-code	1	
	Stroke (0-9)	stroke-cnt (Code:_)	1	3
Attendant Console (Calls Waiting)	CW Aud Off	cw-ringoff	1	
Attendant Control of Trunk Group Access (Activate)	Cont Act	act-tr-grp	1	

Continued on next page

Table 3. Attendant console feature buttons (Continued)

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
Attendant Control of Trunk Group Access (Deactivate)	Cont Deact	deact-tr-g	1	
Attendant Direct Trunk Group Select	Local TG Remote TG	local-tgs remote-tgs	12	4
Attendant Crisis Alert	Crisis Alert	crss-alert	1	
Attendant Display [display buttons]	Date/Time	date-time	1	
	Inspect Mode	inspect	1	
	Normal Mode	normal	1	
	Stored Number	stored-num	1	
Attendant Hundreds Group Select	Group Select _	hundrd-sel (Grp:__)	20 per console	5
Attendant Room Status	Occupied Rooms Status	occ-rooms	1	6
	Maid Status	maid-stat	1	6
Attendant Override	Override	override	1	
Automatic Circuit Assurance	ACA	aca-halt	1 per system	
Automatic Wakeup (Hospitality)	Auto Wakeup	auto-wkup	1	
Busy Verification	Busy Verify	verify	1	
Call Coverage	Cover Cback	cov-cback	1	
	Consult	consult	1	
	Go To Cover	goto-cover	1	
Call Coverage [display button]	Cover Msg Rt	cov-msg-rt	1	
Call Offer (Intrusion)	Intrusion	intrusion	1	

Continued on next page

Table 3. Attendant console feature buttons (Continued)

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
Call Prompting [display button]	Caller Info	callr-info	1	
Call Type	Call Type	type-disp	1	
Centralized Attendant Service	CAS-Backup	cas-backup	1	
Check In/Out (Hospitality) [display buttons]	Check In	check-in	1	
	Check Out	check-out	1	
Class of Restriction [display button]	COR	class-rstr	1	
Conference Display [display button]	Conference Display	conf-dsp	1	
Demand Print	Print Msgs	print-msgs	1	
DID View	DID View	did-view	1	
Do Not Disturb (Hospitality)	Do Not Disturb	dn-dst	1	
Do Not Disturb (Hospitality) [display buttons]	Do Not Disturb Ext	ext-dn-dst	1	
	Do Not Disturb Grp	grp-dn-dst	1	
Don't Split	Don't Split	dont-split	1	
Emergency Access To the Attendant	Emerg. Access To Attd	em-acc-att	1	
Facility Busy Indication [status lamp]	Busy (trunk or extension#)	busy-ind (TAC/Ext: _)	1 per TAC/Ext.	7
Facility Test Calls [status lamp]	FTC Alarm	trk-ac-alm	1	
Far End Mute [display button]	Far End Mute for Conf	fe-mute	1	
Group Display	Group Display	group-disp	1	

Continued on next page

Table 3. Attendant console feature buttons (Continued)

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
Group Select	Group Select	group-sel	1	
Hardware Failure [status lamps]	Major Hdwe Failure	major-alrm	10 per system	
	Auto Wakeup	pr-awu-alm	1	
	DS1 (facility)	ds1-alarm	10 per system	
	PMS Failure	pms-alarm	1	
	PMS Ptr Alm	pr-pms-alm	1	
	CDR 1 Failure	cdr1-alrm	1	
	CDR 2 Failure	cdr2-alrm	1	
	Sys Ptr Alm	pr-sys-alm	1	
Hold	Hold	hold	1	
Integrated Directory [display button]	Integrtd Directory	directory	1	
Incoming Call Identification	Coverage (Group number, type, name, or ext.#)	in-call-id	N	
Intrusion (Call Offer)	Intrusion	intrusion	1	
Leave Word Calling	Cancel LWC	lwc-cancel	1	
	LWC	lwc-store	1	
Leave Word Calling [display buttons]	Delete Msg	delete-msg	1	
	Next	next	1	
	Call Display	call-disp	1	
Leave Word Calling (Remote Message Waiting) [status lamp]	Msg (name or extension #)	aut-msg-wt (Ext:___)	N	
Link Failure	Link Failure (Link No.__)	link-alarm (Link No.__)	1 per Link #	8

Continued on next page

Table 3. Attendant console feature buttons (Continued)

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
Login Security Violation	lsvn-halt	lsvn-halt	1 per system	
Message Waiting	Message Waiting Act.	mwn-act	1 per system	
	Message Waiting Deact.	mwn-deact	1 per system	
Night Service	Trunk Grp. NS	trunk-ns (Grp. No.__)	1 per trunk group	9
PMS Interface [display buttons]	PMS display			
Priority Calling	Prior Call	priority	N	
Position Busy	Position Busy	pos-busy	1	
Queue Status Indications (ACD) [display buttons]	AQC	atd-qcalls	1	
	AQT	atd-qtime		
Queue Status Indications (ACD) [status lamps]	NQC	q-calls (Grp:_)	1	10
	OQT	q-time Grp:_)	1 per hunt group	10
Remote Access Security Violation	rsvn-halt	rsvn-halt	1 per system	
Ringing	In Aud Off	in-ringoff	1	
Security Violation Notification Halt	ssvn-halt	ssvn-halt	1 per system	
Serial Call	Serial Call	serial-cal	1	
Split/Swap	Split-swap	split-swap	1	11
System Reset Alert	System Reset Alert [status lamp]	rs-alert	1	
Station Security Code Notification Halt	ssvn-halt	ssvn-halt	1 per system	

Continued on next page

Table 3. Attendant console feature buttons (Continued)

Feature or Function	Recommended Button Label	Name Entered on Station Screen	Maximum Allowed	Notes
Night Service (ACD)	Hunt Group	hunt-ns (Grp. No.__)	3 per hunt group	12
Time of Day Routing [display buttons]	Immediate Override	man-ovrid	1	
	Clocked Override	clk-overid	1	
Timed Reminder	RC Aud Off	re-ringoff	1	
Timer	Timer	timer	1	
Trunk Identification [display button]	Trunk-ID	trk-id	1	
Trunk Group Name [display button]	Trunk-Name	trunk-name	1	
Visually Impaired Service (VIAS)	VIS	vis	1	
	Console Status	con-stat	1	
	Display	display	1	
	DTGS Status	dtgs-stat	1	
	Last Message	last-mess	1	
	Last Operation	last-op	1	
VDN of Origin Announcement Repeat	VOA Repeat	voa-repeat	1	12
VuStats	VuStats	vu-display	1	

N = any number of buttons on the phone can be assigned to this feature. For phone feature button descriptions, refer to ''Telephone feature buttons'' on page 129.

1. List: List number 1 to 3 where the destination number is stored.
   DC: Dial codes of destination number.
2. Grp: The split group number for ACD.
3. Code: Enter a stroke code (0 through 9).

4. TAC: local-tgs — TAC of local TG
remote-tgs — (L-TAC) TAC of TG to remote PBX
remote-tgs — (R-TAC) TAC of TG on remote PBX
The combination of local-tgs/remote-tgs per console must not exceed 12 (maximum).
Label associated button appropriately so as to easily identify the trunk group.
5. Grp: Enter a hundreds group number (1 through 20).
6. Enhanced Hospitality must be enabled on the System-Parameters Customer-Options screen.
7. Ext: May be a VDN extension.
8. Link: A link number — 1 to 8 for multi-carrier cabinets, 1 to 4 for single-carrier cabinets.
9. Grp: A trunk group number.
10. Grp: Group number of the hunt group.
11. Allows the attendant to alternate between active and split calls.
12. VDN of Origin must be enabled.

Removing an attendant console

Before you physically remove an attendant from your system, check the attendant's status, remove it from any group or usage lists, and then delete it from the system's memory.

For example, to remove attendant 3, which also is assigned extension 4345:

1. Type **status attendant 3** and press RETURN.

   The Attendant Status screen appears.

2. Make sure that the attendant:

   - is plugged into the jack
   - is idle (not making or receiving calls)

3. Type **list usage extension 4345** and press RETURN.

   The Usage screen shows where the extension is used in the system.

4. Press CANCEL.

5. If the attendant extension appears on the Usage screen, access the appropriate feature screen and delete the extension.

   For example, if extension 1234 belongs to hunt group 2, type **change hunt group 2** and delete the extension from the list.

6. Type **remove attendant 3** and press RETURN.

   The system displays the Attendant Console screen so you can verify that you are removing the correct attendant.

7. If this is the correct attendant, press ENTER.

   If the system responds with an error message, the attendant is busy or still belongs to a group. Press CANCEL to stop the request, correct the problem, and enter **remove attendant 3** again.

8. Remove the extension from voice mail service if the extension has a voice mailbox.

9. Type **save translations** and press RETURN to save your changes.

Note that you do not need to delete the extension from coverage paths. The system automatically adjusts coverage paths to eliminate the extension.

Now you can unplug the console from the jack and store it for future use. You do not need to disconnect the wiring at the cross-connect field. The extension and port address remain available for assignment at a later date.

Setting console parameters

You can define system-wide console settings on the Console Parameters screen.

For example, if you want to warn your attendants when there are more than 3 calls in queue or if a call waits for more than 20 seconds, complete the following steps:

1. Type **change console-parameters** and press RETURN.

   The Console Parameters screen appears.

```
change console-parameters                                       Page   1 of   3
                              CONSOLE PARAMETERS
         Attendant Group Name: OPERATORS
                          COS: 0                                     COR: 0
       Calls in Queue Warning: 3                       Attendant Lockout? y
        Ext Alert Port (TAAS):
                          CAS: none
             SAC Notification? n                 Night Service Act. Ext.:
                 IAS (Branch)? n                 IAS Tie Trunk Group No.:
         IAS Att. Access Code:                     Alternate FRL Station:
              Backup Alerting? n           DID-LDN Only to LDN Night Ext? n
TIMING
  Time Reminder on Hold (sec): 10            Return Call Timeout (sec): 10
  Time in Queue Warning (sec): 20
  INCOMING CALL REMINDERS
      No Answer Timeout (sec): 20                         Alerting (sec): 40
                               Secondary Alert on Held Reminder Calls? y
ABBREVIATED DIALING
     List1: group    1        List2:                   List3:

                           COMMON SHARED EXTENSIONS
            Starting Extension:                  Count:
```

2. In the Calls in Queue Warning field, enter **3**.

   The system lights the console's second call waiting lamp if the number of calls waiting in the attendant queue exceeds 3 calls.

3. In the Time in Queue Warning field, enter **20**.

   The system issues a reminder tone if a call waits in the attendant queue for more than 20 seconds.

4. Press ENTER to save changes.

Note that some of the settings on the individual Attendant screens can override your system-wide settings.

Providing backup for an attendant

MultiVantage allows you to configure your system so that you have backup positions for your attendant. Attendant Backup Alerting notifies backup telephones that the attendant need assistance in handling calls. The backup telephones are alerted when the attendant queue reaches the queue warning level or when the console is in night service.

Once a backup telephone receive an alert, the user can dial the Trunk Answer Any Station (TAAS) feature access code to answer the alerting attendant calls.

Tip:
You can find more information about attendant backup in the GuestWorks Technician Handbook.

Before you start

- You can assign the attendant backup alerting only to multiappearance telephones that have a client room class of service (COS) set to No. For more information, refer to ''Class of Service'' on page 650.
- If you have not yet defined a Trunk Answer Any Station (TAAS) feature access code, you need to define one and provide the feature access code to each of the attendant backup users. For more information, refer to ''Feature Access Code'' on page 752.

Instructions

To enable your system to alert backup stations, you need to administer the Console Parameters screen for backup alerting. You also need to give the backup phones an attendant queue calls feature button and train your backup users how to answer the attendant calls.

To configure the system to provide backup alerts and to setup extension 4345 to receive these alerts, complete the following steps:

1. Type **change console-parameters** and press RETURN.

   The Console Parameters screen appears.

2. In the Backup Alerting field, enter **y**.

3. Press ENTER to save changes.

   The system will now notify anyone with an attendant queue calls button when the attendant queue reaches the warning level or when the console is in night service.

4. Type **change station 4345** and press RETURN.

   The Station screen appears.

5. In one of the Button Assignment fields, enter **atd-qcalls**.

   The atd-qcalls button provides the visual alerting for this telephone. When this button is dark (idle state), there are no calls in the attendant queue. When the button shows a steady light (busy state), there are calls in the attendant queue. When button shows a flashing light (warning state), the number of calls in the attendant queue exceeds the queue warning. The backup-telephone user also hears an alerting signal every 10 seconds.

6. Press ENTER to save changes.

Now you need to train the user how to interpret the backup alerting and give them the Trunk Answer Any Station (TAAS) feature access code so that they can answer the attendant calls.

Managing displays

6

Displaying caller information

This chapter provides information on the messages that appear on the read-out screen on display phones.

Your system uses automatic incoming call display to provide information about incoming calls to a display phone that is in use, or active on a call. The information is displayed for 30 seconds on all phones except for CALLMASTER phones, where the display goes blank after 30 seconds. However, the information for each new call overrides the existing message.

Call information appears on the display only if the call terminates at the phone. For example, if the call is forwarded to another extension, no call information appears.

Refer to ''Telephone Displays'' on page 1712 for more information on the buttons and languages you can set up for the messages that appear on the display.

Displaying ANI calling party information

Calling party information may consist of either a billing number that sometimes is referred to as Automatic Number Identification (ANI), or a calling party number. Your phone may display the calling party number and name, or the incoming trunk group name.

Instructions

Let's set up tie trunk group 10 to receive calling party information and display the calling party number on the phone of the person called.

NOTE:
These instructions are for collecting ANI in the U.S. Refer to ''Automatic customer telephone rearrangement'' on page 1376 for information on ANI administration outside the U.S.

1. Type **change trunk group 10**.

   The Trunk Group screen for trunk group 10 appears.

```
                                 TRUNK GROUP

Group Number: 10_                  Group Type: tie_____       CDR Reports: r
  Group Name: Node1 to Node3     _______ COR: 10       TN: ___       TAC: *10__
   Direction: two-way        Outgoing Display? n    Trunk Signaling Type: ____
 Dial Access? n                Busy Threshold: 99_         Night Service: _____
Queue Length: ___                                   Incoming Destination: 2000_
   Comm Type: voice                    Auth Code? n
                                     Trunk Flash? n
         BCC: _                              ITC? ____
TRUNK PARAMETERS
   Trunk Type (in/out): auto/auto-incorrect Incoming Rotary Timeout(sec): 5
    Outgoing Dial Type: _________                  Incoming Dial Type: tone
                                              Disconnect Timing(msec): 500
       Digit Treatment: __________                             Digits: ____
                                                    Sig Bit Inversion: none
    Incoming Dial Tone? y

Disconnect Supervision - In? y  Out? n
  Answer Supervision Timeout: 0__          Receive Answer Supervision? y
```

2. Type **tone** in the Incoming Dial Type field.
3. Press NEXT PAGE and type ***ANI*DNIS** in the Incoming Tone (DTMF) ANI field.
4. Press ENTER to save your changes.

Displaying ICLID Information

Your switch collects the calling party name and number (Incoming Call Line Identification, or ICLID) received from the central office (CO) on analog trunks.

Before you start

Be sure Analog Trunk Incoming Call ID field is set to **y** on the System-Parameters Customer-Options screen.

Refer to the Avaya MultiVantage™ Hardware Solutions Guide for information on the required circuit pack.

Instructions

Let's set up the analog diod trunk group 1 to receive calling party information and display the calling party number on the phone of the person called.

1. Type **change trunk group 1**.

   The Trunk Group screen for trunk group 1 appears. The Group Type field is already set to diod.

2. Press NEXT PAGE to display the Trunk Features page.

```
change trunk-group 1                                                  Page 2 of x
                                  TRUNK GROUP
TRUNK FEATURES
          ACA Assignment? n          Measured: none
                                                         Maintenance Tests? y
                              Data Restriction? n
   Suppress # Outpulsing? n
                                    Receive Analog Incoming Call ID: Bellcore
Incoming Tone (DTMF) ANI: no
```

3. Type **Bellcore** in the Receive Analog Incoming Call ID field.

4. Press NEXT PAGE to display the Administrable Timers section.

```
change trunk-group 1                                                  Page 3 of x
                                  TRUNK GROUP
ADMINISTRABLE TIMERS
   Incoming Disconnect(msec): 500
   Incoming Dial Guard(msec): 70
                                                 Incoming Seizure(msec): 120
          Flash Length(msec): 540  Incoming Incomplete Dial Alarm (sec): 255
END TO END SIGNALING
    Tone (msec): 350   Pause (msec): 150
```

5. Type **120** in the Incoming Seizure (msec) field.

6. Press ENTER to save your changes.

Changing the display language

This section explains how to change the display language.

Before you start

- Make sure the Display Character Set field on the System Parameters Country-Options screen is set to the character type you want to display. This field is set by Avaya.

  NOTE:
  Note: If you change the Display Character Set field to **Roman** after you have administered non-Roman characters, you must change the display field values back to Roman characters on each administrable language display screen. Refer to ''Feature information displays'' on page 1719 for more information.

- Be sure the type of phone your company uses supports the characters you want to display. Each character set requires specific phones. Call your Avaya representative for details.

Instructions

Let's change the display message language to German for the user at attendant console 1, a 40-character display model. Also change the "transfer completed" message from English to German.

1. Type **change attendant 1** and press ENTER.

   The Attendant Console screen appears.

```
change attendant 1                                            Page   1 of   3
                             ATTENDANT CONSOLE 1

        Type: console           Name: Gunther
   Extension: 1000             Group: 1              Auto Answer: none
Console Type: principal           TN: 1              Data Module? n
        Port: 01C1106            COR: 1        Disp Client Redir? n
                                 COS: 1         Display Language: user-defined


DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote            Local Remote             Local Remote
 1: 9                    5:                       9:
 2: 82                   6:                      10:
 3:                      7:                      11:
 4:                      8:                      12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:           5:           9:           13:           17:
 2:           6:          10:           14:           18:
 3:           7:          11:           15:           19:
 4:           8:          12:           16:           20:
```

2. Type **user-defined** in the Display Language field.

   NOTE:
   If "user-defined" is selected for the display language and no translations are defined on the Language Translation screens, all display messages appear as a string of asterisks.

3. Press ENTER to save your changes.
4. Type **change display-language transfer-conference** and press ENTER.

   The Language Translations screen for Transfer Completed appears.

```
change display-language transfer-conference                         Page 1 of 3

                               Language Translations

  1.     English: Transfer completed.
     Translation: abtretung abgeschlossen
```

5. Type **abtretung abgeschlossen** in the Translation field and press ENTER to save your changes.

   Tip:
   To include European, Katakana, or Ukrainian fonts in your display message, use a tilde (~) before and after a Roman character that maps to the character you wish to display. For example, type ~i~ *to create the character* ä *in your German display messages. Refer to* *"Mapping enhanced display characters'' on page 1740* *for character set maps.*

Related topics

Refer to ''Telephone Displays'' on page 1712 more information about choosing the language for messages on your display phones and for mapping US English to Cyrillic (for Russian), Katakana (for Japanese) European, or Ukrainian characters.

Refer to ''System Parameters Country-Options'' on page 1112 for more information about and field descriptions on the System Parameters Country-Option screen.

Fixing problems

Symptom	Cause and Solution
Characters that display are not what you thought you entered.	This feature is case sensitive. Check the table to make sure that you entered the right case.
You entered “~c”, and “*” appears on the display instead.	Lower-case “c” has a specific meaning in the MultiVantage system, and therefore cannot be mapped to any other character. An asterisk “*” appears in its place.
You entered “~->” or “~<-” and nothing appears on the display.	These characters do not exist as single keys on the standard US-English keyboard. Therefore the system is not programmed to handle them.
Enhanced display characters appear in fields that you did not update.	If an existing display field contains a tilde (~) followed by Roman characters, and you update and submit that screen after this feature is activated, that field will display the enhanced character set.
Nothing displays on the terminal at all.	Some unsupported terminals do not display anything if a special character is presented. Check the model of display terminal that you are using.
You entered a character with a descender and part of it appears cut off in the display.	Some of the unused characters in Group2a have descenders that do not appear entirely within the display area. These characters are not included in the character map. For these characters (g,j,p,q,y), use Group1 equivalents.

Setting up directory buttons

Your switch directory contains the names and extensions that are assigned on each station screen. Display-phone users can use a phone button to access the directory, use the touch-tone buttons to key in a name, and retrieve an extension from the directory.

Instructions

Let's assign directory phone buttons for extension 2000. Our button assignment plan is set up so that phone buttons 6, 7, and 8 are used for the directory. Remember, the name you type in the Name field on the first page of the station screen is the name that appears when the directory is accessed on a phone display.

1. Type **change station 2000**.

   The Station screen for extension 2000 appears.

2. Move to the Button Assignments section.

```
                                                                    Page  3 of  X
                                                   STATION
      SITE DATA
           Room: _______                                       Headset? n
           Jack: ___                                           Speaker? n
          Cable: ___                                          Mounting: d
          Floor: _______                                  Cord Length: 0_
       Building: _______                                    Set Color: _____


ABBREVIATED DIALING
     List1: ________             List2: _________            List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                                  5:
 2: call-appr                                  6: directory
 3: call-appr                                  7: next
 4: call-appr                                  8: call-display
```

3. In Button Assignment field 6, type **directory**.
4. In Button Assignment field 7, type **next**.
5. In Button Assignment field 8, type **call-display**.
6. Press ENTER to save your changes.

Handling incoming calls

7

Setting up basic call coverage

This chapter shows you how to set up call coverage for incoming calls to be sure that incoming calls are answered when the called party is not available.

Basic incoming call coverage:

- provides for automatic redirection of calls to alternate destinations when the called party is not available or not accepting calls
- provides the order in which Avaya MultiVantage redirects calls to alternate phones or terminals
- establishes up to 6 alternate termination points for an incoming call
- establishes redirection criteria that govern when a call redirects
- redirects calls to a local telephone number (extension) or an off-switch telephone number (public network)

Call coverage allows an incoming call to redirect from its original destination to an extension, hunt group, attendant group, uniform call distribution (UCD) group, direct department calling (DDC) group, automatic call distribution (ACD) split, coverage answer group, AUDIX, or vector for a station not accepting calls.

Administering system-wide call coverage characteristics

This section shows you how to set up system-wide call coverage characteristics that govern how coverage is handled.

Instructions

The System-Parameters Call Coverage / Call Forwarding screen sets up the global parameters which direct the switch on how to act in certain situations.

1. Leave all default settings as they are set for your system.
2. If you desire to customize your system, carefully read and understand each field description before you make any changes.

   Refer to ‘‘System Parameters Call Coverage / Call Forwarding’’ on page 1104 for more information.

Related topics

Refer to ‘‘Covering calls redirected to an off-site location’’ on page 191 for more information on redirecting calls.

Creating coverage paths

This section explains how to administer various types of call coverage. In general, call coverage refers to what happens to incoming calls. You can administer paths to cover all incoming calls, or define paths for certain types of calls, such as calls to busy phones. You can define where incoming calls go if they are not answered and in what order they reroute to other locations. For example, you can define coverage to ring the called phone, then move to a receptionist if the call is not answered, and finally access a voice mailbox if the receptionist is not available.

With call coverage, the system redirects a call to alternate answering extensions when no one answers at the first extension. An extension can have up to 6 alternate answering points. The system checks each extension in sequence until the call connects. This sequence of alternate extensions is called a coverage path.

The system redirects calls based on certain criteria. For example, you can have a call redirect to coverage without ever ringing on the principal set, or after a certain number of rings, or when one or all call appearances (lines) are busy. You can set coverage differently for internal (inside) and external (outside) calls, and you can define coverage individually for different criteria. For example, you can decide that external calls to busy phones can use the same coverage as internal calls to phones with Do Not Disturb active.

Instructions

To create a coverage path:

1. Type **add coverage path next** and press RETURN.

   The Coverage Path screen appears. The system displays the next undefined coverage path in the sequence of coverage paths. Our example shows coverage path number 2.

```
                              COVERAGE PATH

               Coverage Path Number: 2
                                                Hunt After Coverage: n
                   Next Path Number: ___        Linkage: ___  ___

COVERAGE CRITERIA
    Station/Group Status   Inside Call   Outside Call
             Active?            n             n
               Busy?            y             y
        Don't Answer?           y             y Number of Rings: 3
                All?            n             n
 DND/SAC/Goto Cover?            y             y

COVERAGE POINTS

 Terminate to Coverage Pts. with Bridged Appearance? n

   Point1: _________ Point2: _________ Point3: _________
   Point4: _________ Point5: _________ Point6: _________
```

2. Type a coverage path number in the Next Path field.

   The next path is optional. It is the coverage path to which calls are redirected if the current path's coverage criteria does not match the call status. If the next path's criteria matches the call status, it is used to redirect the call; no other path is searched.

3. Fill in the Coverage Criteria fields.

   You can see that the default sets identical criteria for inside and outside calls. The system sets coverage to take place from a busy phone, if there is no answer after a certain number of rings, or if the DND (do not disturb), SAC (send all calls), or Go to Cover button are pressed or feature-access codes are dialed.

4. Fill in the Point fields with the extensions, hunt group number, or coverage answer group number you want for coverage points.

   Each coverage point can be an extension, hunt group, coverage answer group, remote number, or attendant.

5. Press ENTER to save your changes.

Tip:
If you want to see which extensions or groups use a specific coverage path, type **display coverage sender group n***, where n is the coverage path number. For example, you should determine which extensions use a coverage path before you make any changes to it.*

Assigning a coverage path to users

Now assign the new coverage path to a user. For example, let's assign this new coverage path to extension 2054.

NOTE:
A coverage path can be used for more than one extension.

Instructions

To assign a coverage path:

1. Type **change station 2054** and press RETURN.

   The Station screen for extension 2054 appears.

```
                                    STATION

Extension: 2054                            Lock Messages? n       BCC: 0
     Type: 7406D                           Security Code: ______   TN: 1
     Port: _____                         Coverage Path 1: 2__     COR: 1
     Name: __________________________    Coverage Path 2: ___     COS: 1
                                         Hunt-to-Station: _______

STATION OPTIONS
             Data Module? n              Personalized Ringing Pattern: 1
          Display Module? n                         Message Lamp Ext: ________


                                                 MM Complex Data Ext: ________
```

2. Type **2** in the Coverage Path 1 field.

   To give extension 2054 another coverage path, you can type a coverage path number in the Coverage Path 2 field.

3. Press ENTER to save your changes.

Related information

''Assigning coverage options'' on page 357

Setting up advanced call coverage

Advanced incoming call coverage:

- redirects calls based on time-of-day
- allows coverage of calls that are redirected to sites not on the local switch
- allows users to change back and forth between two coverage choices (either specific lead coverage paths or time-of-day tables).

Covering calls redirected to an off-site location

You can provide coverage for calls that have been redirected to an off-site location (for example, your home). This capability, called Coverage of Calls Redirected Off-Net (CCRON) allows you to redirect calls onto the public network and bring back unanswered calls for further coverage processing.

Before you start

- On the System-Parameters Customer-Options screen, verify the Coverage of Calls Redirected Off-Net Enabled field is **y**. If not, contact your Avaya representative.
- You need call classifier ports for all situations except ISDN end-to-end signaling, in which case the ISDN protocol does the call classification. For all other cases, use one of the following:
  - Tone Clock with Call Classifier - Tone Detector circuit pack. Refer to Avaya MultiVantage™ Hardware Solutions Guide for more information on the circuit pack.
  - Call Classifier - Detector circuit pack.

Instructions

To provide coverage of calls redirected to an off-site location:

1. Type **change system-parameters coverage-forwarding** and press RETURN.

   The System Parameters Call Coverage / Call Forwarding screen appears.

```
                SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)
                            Coverage of Calls Redirected Off-Net Enabled? y
      Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? y
                                       Ignore Network Answer Supervision? n
         Immediate Redirection On Receipt Of PROGRESS Inband Information? n
```

2. In the Coverage of Calls Redirected Off-Net Enabled field, type **y**.

   This instructs Avaya MultiVantage to monitor the progress of an off-net coverage or off-net forwarded call and provide further coverage treatment for unanswered calls.

3. In the Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point field, leave the default as **y**.
4. In the Ignore Network Answer Supervision field, leave the default as **n**.
5. In the Immediate Redirection On Receipt Of PROGRESS Inband Information field, leave the default as **n**.
6. Press ENTER to save your changes.

Defining coverage for calls redirected to external numbers

You can administer the system to allow calls in coverage to redirect to off-net (external) or public-network numbers.

Standard remote coverage to an external number allows you to send a call to an external phone, but does not monitor the call once it leaves your system. Therefore, if the call is busy or not answered at the external number, the call cannot be pulled back to the system. With standard remote call coverage, make the external number the last coverage point in a path.

With newer systems, you may have the option to use the Coverage of Calls Redirected Off-Net feature. If this feature is active and you use an external number in a coverage path, the system can monitor the call to determine whether the external number is busy or does not answer. If necessary, the system can redirect a call to coverage points that follow the external number. With this feature, you can have a call follow a coverage path that starts at the user’s extension, redirects to the user’s home phone, and if not answered at home, returns to redirect to their voice mail box.

The call will not return to the system if the external number is the last point in the coverage path.

To use a remote phone number as a coverage point, you need to define the number in the Remote Call Coverage Table and then use the remote code in the coverage path.

Instructions

For example, to add an external number (303-538-1000) to coverage path 2:

1. Type **change coverage remote** and press RETURN.

   The Remote Call Coverage Table appears.

```
                         REMOTE CALL COVERAGE TABLE
01: 93035381000_____       16: ________________      31: ________________
02: ________________       17: ________________      32: ________________
03: ________________       18: ________________      33: ________________
04: ________________       19: ________________      34: ________________
05: ________________       20: ________________      35: ________________
06: ________________       21: ________________      36: ________________
07: ________________       22: ________________      37: ________________
08: ________________       23: ________________      38: ________________
09: ________________       24: ________________      39: ________________
10: ________________       25: ________________      40: ________________
11: ________________       26: ________________      41: ________________
12: ________________       27: ________________      42: ________________
13: ________________       28: ________________      43: ________________
14: ________________       29: ________________      44: ________________
15:________________        30: ________________      45: ________________
```

2. Type **93035381000** in one of the remote code fields.

   If you use a digit to get outside of your network, you need to add the digit before the external number. In this example, the system requires a ‘9’ to place outside calls.

3. Be sure to record the remote code number you use for the external number.

   In this example, the remote code is r01.

4. Press ENTER to save your changes.

5. Type **change coverage path 2** and press RETURN.

   The Coverage Path screen appears.

   Tip:
   Before making changes, you can use ***display coverage sender group 2*** *to determine which extensions or groups use path 2.*

```
                        COVERAGE PATH

                     Coverage Path Number: _
                                                Hunt After Coverage: y
                         Next Path Number: ___  Linkage: ___   ___

COVERAGE CRITERIA
    Station/Group Status   Inside Call   Outside Call
             Active?            n             n
               Busy?            y             y
       Don't Answer?            y             y Number of Rings: 2
                All?            n             n
 DND/SAC/Goto Cover?            y             y

COVERAGE POINTS

 Terminate to Coverage Pts. with Bridged Appearance? n

              Point1: 4101_____ Point2: r1_______ Point3: h77_____
              Point4: _________ Point5: _________ Point6: _________
```

6. Type r1 in a coverage Point field.

   In this example, the coverage rings at extension 4101, then redirects to the external number. If you administer Coverage of Calls Redirected Off-Net and the external number is not answered or is busy, the call redirects to the next coverage point. In this example, the next point is Point 3 (h77 or hunt group 77).

   If you do not have the Coverage of Calls Redirected Off-Net feature, the system cannot monitor the call once it leaves the network. The call ends at the remote coverage point.

7. Press ENTER to save your changes.

Related topics

Refer to ''Call Coverage'' on page 1416 for more information on coverage.

Defining time-of-day coverage

The Time of Day Coverage Table on your system lets you redirect calls to coverage paths according to the time of day and day of the week when the call arrives. You need to define the coverage paths you want to use before you define the time of day coverage plan.

For example, let's say you want to administer the system so that incoming calls to extension 2054 redirect to a coworker in the office from 8:00 a.m. to 5:30 p.m., and to a home office from 5:30 p.m. to 8:00 p.m. on weekdays. You want to redirect the calls to voice mail after 8:00 p.m. weekdays and on weekends.

Instructions

To set up a time-of-day coverage plan that redirects calls for our example above:

1. Type **add coverage time-of-day next** and press RETURN.

   The Time of Day Coverage Table screen appears and the selects the next undefined table number in the sequence of time-of-day table numbers. If this is the first time-of-day coverage plan in your system, the table number is 1.

```
                         TIME OF DAY COVERAGE TABLE 1__

           Act    CVG      Act    CVG      Act    CVG      Act    CVG      Act    CVG
           Time   PATH     Time   PATH     Time   PATH     Time   PATH     Time   PATH

  Sun      00:00 3___      __:__ ____      __:__ ____      __:__ ____      __:__ ____
  Mon      00:00 3___      08:00 1         17:30 2         20:00 3         __:__ ____
  Tue      00:00 3___      08:00 1         17:30 2         20:00 3         __:__ ____
  Wed      00:00 3___      08:00 1         17:30 2         20:00 3         __:__ ____
  Thu      00:00 3___      08:00 1         17:30 2         20:00 3         __:__ ____
  Fri      00:00 3___      08:00 1         17:30 2         20:00 3         __:__ ____
  Sat      00:00 3___      __:__ ____      __:__ ____      __:__ ____      __:__ ____
```

   Record the table number so that you can assign it to extensions later.

2. To define your coverage plan, enter the time of day and path number for each day of the week and period of time.

   Enter time in a 24-hour format from the earliest to the latest. For this example, assume that coverage path 1 goes to the coworker, path 2 to the home, and path 3 to voice mail.

   Define your path for the full 24 hours (from 00:01 to 23:59) in a day. If you do not list a coverage path for a period of time, the system does not provide coverage for that time.

3. Press ENTER to save your changes.

Now assign the time-of-day coverage to a user. For example, we use extension 2054:

1. Type **change station 2054** and press RETURN.

   The Station screen for extension 2054 appears.

```
                                     STATION

Extension: 2054                               Lock Messages? n       BCC: 0
     Type: 7406D                              Security Code: 4196__   TN: 1
     Port: _____                            Coverage Path 1: t1_     COR: 1
     Name: ___________________________      Coverage Path 2: ___     COS: 1
                                            Hunt-to-Station: _______

STATION OPTIONS
             Data Module? n                 Personalized Ringing Pattern: 1
          Display Module? n                            Message Lamp Ext: ________


                                                     MM Complex Data Ext: ________
```

2. Move your cursors to Coverage Path 1 and type **t** plus the number of the Time of Day Coverage Table.
3. Press ENTER to save your changes.

Now calls to extension 2054 redirect to coverage depending on the day and time that each call arrives.

Creating coverage answer groups

You can create a coverage answer group so that up to 8 phones simultaneously ring when calls cover to the group. Anyone in the answer group can answer the incoming call.

Instructions

To add a coverage answer group:

1. Type **add coverage answer-group next** and press RETURN.

   The Coverage Answer Group screen appears.

```
                          COVERAGE ANSWER GROUP

                       Group Number: ___
                       Group Name: COVERAGE_GROUP_

GROUP MEMBER ASSIGNMENTS
   Ext      Name (first 26 characters)      Ext      Name (first 26 characters)
1: ________ ___________________________ 5: ________ __________________________
2: ________ ___________________________ 6: ________ __________________________
3: ________ ___________________________ 7: ________ __________________________
4: ________ ___________________________ 8: ________ __________________________
```

2. In the Group Name field, enter a name to identify the coverage group.
3. In the Ext field, type the extension of each group member.
4. Press ENTER to save your new group list.

   The system automatically completes the Name field when you press ENTER.

Related topics

Refer to ''Assigning a coverage path to users'' on page 190 for instructions on assigning a coverage path.

Setting up call forwarding

This section explains how to administer various types of automatic call forwarding. To provide call forwarding to your users, assign each extension a Class of Service (COS) that allows call forwarding. Then assign call-forwarding buttons to the user phones (or give them the Feature Access Code (FAC) for call forwarding) so that they can easily forward calls. You use the station screen to assign the COS and any call-forwarding buttons.

Within each class of service, you can determine whether the users in that COS have the following call forwarding features:

- Call Forwarding All Calls — allows users to redirect all incoming calls to an extension, attendant, or external phone number.
- Call Forwarding Busy/Don't Answer — allows users to redirect calls only if their extensions are busy or they do not answer.
- Restrict Call Fwd-Off Net — prevents users from forwarding calls to numbers that are outside your system network.

As the administrator, you can administer system-wide call-forwarding parameters to control when calls are forwarded. Use the System Parameters Call Coverage/Call Forwarding screen to set the number of times an extension rings before the system redirects the call because the user did not answer (CFWD No Answer Interval). For example, if you want calls to ring 4 times at an extension and, if the call is not answered, redirect to the forwarding number, set this parameter to 4.

You also can use the System Parameters Call Coverage/Call Forwarding screen to determine whether the forwarded-to phone can override call forwarding to allow calls to the forwarded-from phone (Call Forward Override). For example, if an executive forwards incoming calls to an attendant and the attendant needs to call the executive, the call can be made only if the Call Forwarding Override is set to yes.

Instructions

To determine which extensions have call forwarding activated:

1. Type **list call-forwarding** and press RETURN.

   This command lists all the extensions that are forwarded along with each forwarding number.

   NOTE:
   If you have a V1, V2, or V3 system, you can see if a specific extension is forwarded only by typing **status station nnnn**, where nnnn is the specific extension.

Related topics

''Call Forwarding'' on page 1496

Setting up call forwarding for users

This section shows you how to give your users access to call forwarding.

Instructions

Let's change a call forwarding access code from a local phone with a Class of Service of 1:

1. Type **change feature-access-codes** and press RETURN.

   The Feature Access Code screen appears.

```
                                FEATURE ACCESS CODE (FAC)
          Abbreviated Dialing List1 Access Code: ____
          Abbreviated Dialing List2 Access Code: ____
          Abbreviated Dialing List3 Access Code: ____
 Abbreviated Dial - Prgm Group List Access Code: ____
                       Announcement Access Code: ____
                        Answer Back Access Code: ____
       Auto Alternate Routing (AAR) Access Code: ____
       Auto Route Selection (ARS) Access Code 1: ____   Access Code 2: ____
                  Automatic Callback Activation: ____    Deactivation: ____
 Call Forwarding Activation Busy/DA: *70    All: *71     Deactivation: #72
                          Call Park Access Code: ____
                        Call Pickup Access Code: ____
 CAS Remote Hold/Answer Hold-Unhold Access Code: ____
                 CDR Account Code Access Code: ____
                  Change Coverage Access Code: ____
                 Data Origination Access Code: ____
                     Data Privacy Access Code: ____
                Directed Call Pickup Access Code: ____
      Emergency Access to Attendant Access Code: ____
  Extended Call Fwd Activate Busy D/A: ____ All: ____    Deactivation: ____
```

2. In the Call Forwarding Activation Busy/DA field, type ***70**.

   The ***70** feature access code activates the call forwarding option so incoming calls forward when your phone is busy or does not answer.

3. In the Call Forwarding Activation All field, type ***71**.

   The ***71** feature access code forwards all calls.

4. In the Call Forwarding Deactivation field, type **#72**.

   The **#72** feature access code deactivates the call forwarding option.

5. Press ENTER to save your changes.

6. Type **change cos** and press RETURN.

   The Class of Service screen appears.

```
                          CLASS OF SERVICE

                                  0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15

      Auto Callback               n y y n y n y n y n  y  n  y  n  y  n
      Call Fwd-All Calls          n y n y y n n y y n  n  y  y  n  n  y
      Data Privacy                n y n n n y y y y n  n  n  n  y  y  y
      Priority Calling            n y n n n n n n n y  y  y  y  y  y  y
      Console Permissions         n n n n n n n n n n  n  n  n  n  n  n
      Off-hook Alert              n n n n n n n n n n  n  n  n  n  n  n
      Client Room                 n n n n n n n n n n  n  n  n  n  n  n
      Restrict Call Fwd-Off Net   n y y y y y y y y y  y  y  y  y  y  y
      Call Forward Busy/DA        n y n n n n n n n n  n  n  n  n  n  n
      Personal Station Access     n n n n n n n n n n  n  n  n  n  n  n
      Extended Forwarding All     n n n n n n n n n n  n  n  n  n  n  n
      Extended Forwarding B/DA    n n n n n n n n n n  n  n  n  n  n  n
      Trk-to-Trk Restriction Override n n n n n n n n n n  n  n  n  n  n  n
      QSIG Call Offer Originations    n n n n n n n n n n  n  n  n  n  n  n
```

7. On the Call Fwd-All Calls line, in the 1 column, type **y**.

   This allows the user with this Class of Service to forward their calls. The "1" column is for phones with a Class of Service of 1.

8. On the Restrict Call Fwd-Off Net line, in the 1 column, type **y**.

   This restricts your users from forwarding calls off-site. If you want your users to be able to call off-site, leave this field as **n**.

9. On the Call Forward Busy/DA line, in the 1 column, type **y**

   This forwards your calls when your phone is busy or doesn't answer after a programmed number of rings.

10. Press ENTER to save your changes.

Allowing users to specify a forwarding destination

Now that you have set up system-wide call forwarding, have your users use this procedure if they want to change their call forwarding destination from their work (local) station.

1. They dial either their Call Forwarding Activation Busy/DA or Call Forwarding Activation All feature access code. If your users have buttons assigned, they press those buttons, listen for dial tone, and dial the digits.

   NOTE:
   Both Call Forwarding Activation Busy/DA or the Call Forwarding Activation All cannot be active for the same phone at the same time.

   In this example, enter ***71** for Call Forwarding Activation All.

2. They dial their "forwarding-to" off-site or on-site number.

   In this example, enter **2081**. This is a local number; for offsite forwarding, include the AAR/ARS feature access code.

3. When they hear the 3-beep confirmation tone, they hang up.

Changing the forwarding destination remotely

Now that you have set up all of the required system administration for call forwarding, have your users use this procedure if they want to change their call forwarding destination from a telecommuting (off-site) phone.

1. They dial their telecommuting extension.

   Refer to ''Telecommuting Access'' on page 1148 for more information. In this example, enter **555-9126**.

2. When they get dial tone, they dial either their Extended Call Forward Activate Busy/DA or the Extended Call Forward Activate All feature access code.

   In this example, enter ***61** for the Extended Call Forward Activate All number.

3. When they get dial tone, they dial their extension number and press the '**#**' key.

   In this example, enter **1014**, then **#**.

4. Even though there is no dial tone, they dial their security code and press the '**#**' key.

   In this example, enter **4196**, then **#**.

5. When they get dial tone, they dial their "forwarding-to" off-site or on-site number.

   In this example, enter **9-555-2081**.

6. When they hear the 3-beep confirmation tone, they hang up.

Allowing users to change coverage remotely

This section shows you how to allow users to change their call coverage path from a local or telecommuting (off-site) phone.

Instructions

To change call coverage from off-site:

1. Type **change feature-access-codes** and press RETURN.

   The Feature Access Code screen appears.

2. In the Change Coverage Access Code field, type ***85**.

   Use the ***85** feature access code to change a coverage path from a phone or remote station.

3. Press ENTER to save your changes.

4. Type **change cor** and press RETURN.

   The Class of Restriction screen appears.

5. In the Can Change Coverage field, type **y**.

   This permits users to select one of two previously administered coverage paths.

6. Press ENTER to save your changes.

7. Type **change station 1014** and press RETURN.

   The Station screen for extension 1014 appears.

8. In the Security Code field, type **4196**.

   In this example, this is your security code. Refer to ''Security-Related System Parameters'' on page 1047 for more information on setting the length of your security code.

9. In the Coverage Path 1 and Coverage Path 2 fields, verify that both are defined enabling your user to move from one coverage path to another.

   The **t1** and **t2** are the numbers of the Time of Day Coverage Tables.

10. Press ENTER to save your changes.

Setting up night service

You can use night service to direct calls to an alternate location when the primary answering group is not available. For example, you can administer night service so that anyone in your marketing department can answer incoming calls when the attendant is at lunch or has left for the day.

Once you administer night service to route calls, your end-users merely press a button on the console or a feature button on their phones to toggle between normal coverage and night service.

There are five types of night service:

- Night Console Night Service — directs all attendant calls to a night or day/night console
- Night Station Night Service — directs all incoming trunk or attendant calls to a night service destination
- Trunk Answer from Any Station (TAAS) — directs incoming attendant calls and signals a bell or buzzer to alert other employees that they can answer the calls
- Trunk Group Night Service — directs incoming calls to individual trunk groups to a night service destination
- Hunt Group Night Service — directs hunt group calls to a night service destination

Setting up night station service to voice mail

The night station service (also known as Listed Directory Number (LDN) Night Service) sends calls directed to an LDN to voice mail when the system is in night service.

Instructions

What is described below is a common setup; however, you can use a regular extension in this field, but it will not follow coverage.

> **NOTE:**
> You can use a dummy hunt group (one with no members) or an exported station with a coverage path. The instructions below use a hunt group.

To set up a night station service to voice mail:

1. Type **add hunt-group next** and press RETURN.

   The Hunt Group screen appears.

```
                                                              Page 1 of X
                                    HUNT GROUP
         Group Name: ldn nights
       Group Number: 5                Group Extension: 51002___  Group Type:
   MM Early Answer? _                              Skill? _             ACD? _
             Queue? _                             Vector? _             AAS? _
      Security Code: ____                                               COR: _
  ISDN Caller Disp: ________                                             TN: _

                Measured: ________               Supervisor: Extension: ________

       Controlling Adjunct: ____
   Multiple Call Handling: __________
                 Objective: ____

              Queue Length: ___
 Calls Warning Threshold: ___                 Calls Warning Port: __
  Time Warning Threshold: ___                  Time Warning Port: __

                     Redirect on No Answer (rings): _  Redirect to VDN: _
                Forced Entry of Stroke Counts or Call Work Codes? _
```

   The Group Number field fills automatically with the next hunt group number.

2. In the Group Name field, type the name of the group.

   In our example, type **ldn nights**. There should be no members in this hunt group.

3. Press ENTER to save your changes.

   NOTE:
   If you are using tenant partitioning, the command for the next step will be **change tenant x**. If you are using tenant partitioning, the Night Destination field does not appear on the Listed Directory Numbers screen. Instead, it is on the Tenant screen.

4. Type **change listed-directory-numbers** and press RETURN.

   The Listed Directory Numbers screen appears.

```
                                                                Page    1 of    2
                          LISTED DIRECTORY NUMBERS


              Ext      Name                             TN
         1: 51001    Attendant                           1
         2:                                              1
         3:                                              1
         4:                                              1
         5:                                              1
         6:                                              1
         7:                                              1
         8:                                              1
         9:                                              1
        10:                                              1


                         Night Destination: 51002
```

5. In the Night Destination field, add the night destination on the listed directory phone.

   In our example, type **51002**.

6. Press ENTER to save your changes.

7. Type **change console-parameters** and press RETURN.

   The Console Parameters screen appears.

```
                               CONSOLE PARAMETERS
         Attendant Group Name: 27 character name  OPERATOR
                          COS: 1                                  COR: 1
       Calls in Queue Warning: 5                    Attendant Lockout? y
        Ext Alert Port (TAAS):
                          CAS: none
             SAC Notification? n            Night Service Act. Ext.: 1234
                 IAS (Branch)? n            IAS Tie Trunk Group No.:
         IAS Att. Access Code:                Alternate FRL Station:
              Backup Alerting? n       DID-LDN Only to LDN Night Ext? n
TIMING
  Time Reminder on Hold (sec): 10          Return Call Timeout (sec): 10
  Time in Queue Warning (sec):
  INCOMING CALL REMINDERS
      No Answer Timeout (sec): 20                      Alerting (sec): 40
                             Secondary Alert on Held Reminder Calls? y
ABBREVIATED DIALING
     List1: group    1        List2:                  List3:

                            COMMON SHARED EXTENSIONS
            Starting Extension:                  Count:
```

8. In the DID-LDN Only to LDN Night Extension field, type **n**.

9. Press ENTER to save your changes.

10. From a phone with console permissions, dial the call forwarding feature access code, then the hunt group’s extension, followed by the main number of AUDIX.

    In our example, dial 51002.

    NOTE:
    You should receive the confirmation tone (3 beeps). This step is very important as calls to the LDN night service extension do not follow coverage.

11. In voice mail, build your auto attendant with the extension of the Listed Directory Number, not the hunt group.

    The originally dialed number was the LDN. That is what the switch passes to the voice mail. In the case of the Intuity and newer embedded AUDIX Voice Mail systems, you can use the Auto Attendant routing table to send the calls to a common Auto Attendant mailbox.

Setting up night console service

This section shows you how to set up night console service.

Night Console Service directs all calls for primary and daytime attendant consoles to a night console. When a user activates Night Console Service, the Night Service button for each attendant lights and all attendant-seeking calls (and calls waiting) in the queue are directed to the night console.

NOTE:
Activating night console service also puts trunk groups into night service, except those for which a night service button has been administered. Refer to ‘‘Setting up trunk answer from any station’’ on page 209 for more information.

To activate and deactivate Night Console Service, the attendant typically presses the NIGHT button on the principal attendant console or designated console.

Only the principal console can activate night service. In the absence of any console, a phone can activate night service.

Instructions

Let's put the attendant console (attendant 2) in a night service mode.

To set up Night Console Service:

1. Type **change attendant 2** and press RETURN.

   The Attendant Console screen appears.

```
                              ATTENDANT CONSOLE 2

        Type: console           Name: 27 character attd cons name
   Extension: 1000             Group: 1                Auto Answer: none
Console Type: principal           TN: 1                Data Module? n
        Port: 01C1106            COR: 1          Disp Client Redir? n
                                 COS: 1           Display Language: english


DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote           Local Remote              Local Remote
 1: 9                   5:                        9:
 2: 82                  6:                       10:
 3:                     7:                       11:
 4:                     8:                       12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:            5:            9:           13:           17:
 2:            6:           10:           14:           18:
 3:            7:           11:           15:           19:
 4:            8:           12:           16:           20:
```

2. In the Console Type field, type **principal**.

   There can be only one night-only or one day/night console in the system unless you administer Tenant Partitioning. Night Service is activated from the principal console or from the one station set per-system that has a nite-serv button.

3. Press ENTER to save your changes.

Setting up night station service

You can use night station service if you want to direct incoming trunks calls, DID-LDN (direct inward dialing-listed directory number) calls, or internal calls to the attendant (dialed 'O' calls) to a night service destination.

Let's say your attendant, who answers extension (LDN) 8100, usually goes home at 6:00 p.m. When customers call extension 8100 after hours, you would like them to hear an announcement that asks them to try their call again in the morning.

To set up night station service, you need to record the announcement (in our example, it is recorded at announcement extension 1234). See “Managing announcements’’ on page 431 for information on setting up the announcement.

Tip:

All trunk groups that are routed through the attendant direct to this night service destination provided they already do not have a night service destination and, on the Console Parameters screen, the DID-LDN Only to DID-LDN Night Ext field is ***n****. Refer to* *“Setting up trunk answer from any station’’ on page 209**.*

Instructions

To set up night station service:

1. Type **change listed-directory-numbers** and press RETURN.

   The Listed Directory Numbers screen appears.

```
                              LISTED DIRECTORY NUMBERS


          Ext      Name                           TN
     1: 8100     attendant 8100
     2:
     3:
     4:
     5:
     6:
     7:
     8:
     9:
    10:


                                                   Night Destination: 1234
```

2. Enter **1234** in the Night Destination field.

   The destination can be an extension, a recorded announcement extension, a vector directory number, or a hunt group extension.

3. Press ENTER to save your changes.
4. Type **change console-parameters** and press RETURN.

   The Console Parameters screen appears.

```
                              CONSOLE PARAMETERS
         Attendant Group Name: 27 character name  OPERATOR
                          COS: 1                                        COR: 1
       Calls in Queue Warning: 5                    Attendant Lockout? y
        Ext Alert Port (TAAS):
                          CAS: none
             SAC Notification? n              Night Service Act. Ext.: 1234
                 IAS (Branch)? n              IAS Tie Trunk Group No.:
         IAS Att. Access Code:                  Alternate FRL Station:
              Backup Alerting? n        DID-LDN Only to LDN Night Ext? n
TIMING
  Time Reminder on Hold (sec): 10           Return Call Timeout (sec): 10
  Time in Queue Warning (sec):
  INCOMING CALL REMINDERS
      No Answer Timeout (sec): 20                       Alerting (sec): 40
                               Secondary Alert on Held Reminder Calls? y
ABBREVIATED DIALING
     List1: group    1        List2:                  List3:

                           COMMON SHARED EXTENSIONS
            Starting Extension:                  Count:
```

5. In the DID-LDN Only to LDN Night Extension field, type **n**.
6. Press ENTER to save your changes.

   After you set up night station service, have the attendant use the night console button to activate and deactivate night service.

Setting up trunk answer from any station

There may be situations where you want everyone to be able to answer calls when the attendant is away. Use trunk answer any station (TAAS) to configure the system so that it notifies everyone when calls are ringing. Then, you can give users the trunk answer any station feature access code so they can answer these calls.

When the system is in night service mode, attendant calls redirect to an alerting device such as a bell or a buzzer. This lets other people in the office know when they should answer the phone.

NOTE:
If no one answers the call, the call will not redirect to night service.

Let’s define a feature access code (we’ll use 71) and configure the alerting device for trunk answer any station.

Before you start

You need a ringing device and 1 port on an analog line circuit pack. Refer to Avaya MultiVantage™ Hardware Solutions Guide for more information on the circuit pack.

Instructions

To set the feature access code for TAAS:

1. Type **change feature-access-codes** and press RETURN.

   The Feature Access Code screen appears.

```
                                                        Page 3 of X
                            FEATURE ACCESS CODE (FAC)
      Station Security Code Change Access Code: ____
             Terminal Dial-up Test Access Code: ____
Terminal Translation Initialization Merge Code: ____  Separation Code: ____
                 Transfer to AUDIX Access Code: ____
          Trunk Answer Any Station Access Code: 71__
              User Control Restrict Activation: ____ Deactivation: ____
  Voice Coverage Message Retrieval Access Code: ____
 Voice Principal Message Retrieval Access Code: ____
```

2. In the Trunk Answer Any Station Access Code field, type **71**.
3. Press ENTER to save your changes.

Once you set the feature access code, determine where the external alerting device is connected to the switch (we'll use port 01A0702).

To set up external alerting:

1. Type **change console-parameters** and press RETURN.

   The Console Parameters screen appears.

```
                               CONSOLE PARAMETERS
         Attendant Group Name: Operator
                          COS: 0                                   COR: 0
       Calls in Queue Warning: 5                     Attendant Lockout? y
        Ext Alert Port (TAAS): 01A0702
                          CAS: none
             SAC Notification? n               Night Service Act. Ext.:
                 IAS (Branch)? n               IAS Tie Trunk Group No.:
         IAS Att. Access Code:                   Alternate FRL Station:
              Backup Alerting? n          DID-LDN Only to LDN Night Ext? n
TIMING
  Time Reminder on Hold (sec): 10           Return Call Timeout (sec): 10
  Time in Queue Warning (sec):
  INCOMING CALL REMINDERS
      No Answer Timeout (sec): 20                      Alerting (sec): 40
                               Secondary Alert on Held Reminder Calls? y
ABBREVIATED DIALING
     List1: group    1        List2:                  List3:

                            COMMON SHARED EXTENSIONS
            Starting Extension:                 Count:
```

2. In the EXT Alert Port (TAAS) field, type **01A0702**.

   Use the port address assigned to the external alerting device.

3. Press ENTER to save your changes.

Setting up external alerting night service

Calls redirected to the attendant via Call Forwarding or Call Coverage will not go to the Listed Directory Number (LDN) Night Station. If there is no night station specified, and the TAAS bell is being used, these calls ring the TAAS bell. A call following the coverage path rings the TAAS bell for the number of times indicated in the Coverage Don't Answer Interval for Subsequent Redirection (Rings) field. If not answered, the call proceeds to the next point in the station's coverage path. If the call was sent to the Attendant by Call Forwarding, it continues to ring the TAAS bell.

When night service is enabled, and there is a night service destination on the LDN screen, calls covering to the attendant attempt to ring the night destination instead of the attendant position even if the handset is plugged in.

Instructions

To send LDN calls to the attendant during the day and to a guard's desk at night:

1. Type **change listed-directory-numbers** and press RETURN.

   The Listed Directory Numbers screen appears.

```
                                                           Page   1 of   2
                         LISTED DIRECTORY NUMBERS


          Ext      Name                            TN
     1: 2000     Attendant                      1
     2:                                         1
     3:                                         1
     4:                                         1
     5:                                         1
     6:                                         1
     7:                                         1
     8:                                         1
     9:                                         1
    10:                                         1


                         Night Destination: 3000
```

2. In the Night Destination field, verify this field is blank.

3. Press ENTER to save your changes.

4. Type **change console-parameters** and press RETURN.

   The Console Parameters screen appears.

```
                              CONSOLE PARAMETERS
         Attendant Group Name: Operator
                         COS: 0                                COR: 0
       Calls in Queue Warning: 5                  Attendant Lockout? y
        Ext Alert Port (TAAS): 01A0702
                         CAS: none
             SAC Notification? n            Night Service Act. Ext.:
                 IAS (Branch)? n            IAS Tie Trunk Group No.:
         IAS Att. Access Code:                Alternate FRL Station:
              Backup Alerting? n       DID-LDN Only to LDN Night Ext? n
TIMING
  Time Reminder on Hold (sec): 10          Return Call Timeout (sec): 10
  Time in Queue Warning (sec):
  INCOMING CALL REMINDERS
      No Answer Timeout (sec): 20                      Alerting (sec): 40
                               Secondary Alert on Held Reminder Calls? y
ABBREVIATED DIALING
     List1: group    1        List2:                   List3:

                           COMMON SHARED EXTENSIONS
            Starting Extension:                  Count:
```

5. In the EXT Alert Port (TAAS) field, type **01A0702**.

   This is the port address assigned to the external alerting device.

6. Press ENTER to save your changes.

   The system is in Night Service.

   Any calls to extension 2000 now go to extension 3000 (the guard’s desk).

   Any “0” seeking calls go to extension 3000 (the guard’s desk).

To send LDN calls to the attendant during the day and to the TAAS bell at night:

1. Type **change console-parameters** and press RETURN.

   The Console Parameters screen appears.

```
                              CONSOLE PARAMETERS
         Attendant Group Name: Operator
                         COS: 0                              COR: 0
       Calls in Queue Warning: 5                Attendant Lockout? y
        Ext Alert Port (TAAS): 01A0702
                         CAS: none
             SAC Notification? n            Night Service Act. Ext.:
                 IAS (Branch)? n            IAS Tie Trunk Group No.:
         IAS Att. Access Code:                Alternate FRL Station:
              Backup Alerting? n       DID-LDN Only to LDN Night Ext? y
TIMING
  Time Reminder on Hold (sec): 10          Return Call Timeout (sec): 10
  Time in Queue Warning (sec):
  INCOMING CALL REMINDERS
      No Answer Timeout (sec): 20                      Alerting (sec): 40
                               Secondary Alert on Held Reminder Calls? y
ABBREVIATED DIALING
     List1: group    1        List2:                  List3:

                           COMMON SHARED EXTENSIONS
            Starting Extension:                  Count:
```

2. In the DID-LDN Only to Night Ext. field, type **y**.

   This allows only listed directory number calls (LDN) to go to the listed directory night service number extension.

3. In the Ext Alert Port (TAAS) field, type **01A070**.

   This is the port address assigned to the external alerting device.

4. Press ENTER to save your changes.

   The system is in night service.

   Any DNIS extension 2000 calls now go to the TAAS bell.

   Any “0” seeking calls now go to the TAAS bell.

Setting up trunk group night service

You can use trunk group night service if you want to direct individual trunk groups to night service. The system redirects calls from the trunk group to the group’s night service destination.

Trunk group night service overrides night station service. For example, let’s say you activate trunk group night service, and then your attendant activates night station service. In this case, calls to the trunk group use the trunk night service destination, rather than the station night service destination.

Instructions

Let's direct night calls for trunk group 2 to extension 1245.

To set up trunk group night service:

1. Type **change trunk-group 2** and press RETURN.

   The Trunk Group screen appears.

```
                              TRUNK GROUP

Group Number: 2                    Group Type: co          CDR Reports: y
  Group Name: outside calls    COR: 1_        TN: 1__      TAC: ____
   Direction: two-way_       Outgoing Display? n
 Dial Access? n                Busy Threshold: 99      Night Service: 1245
Queue Length: 0                  Country: 1_    Incoming Destination: ________
   Comm Type: voice              Auth Code? n  Digit Absorption List: _
    Prefix-1? y                   Trunk Flash? n     Toll Restricted? y
         BCC: _
TRUNK PARAMETERS
            Trunk Type: loop-start
    Outgoing Dial Type: tone                                Cut-Through? n
     Trunk Termination: rc                     Disconnect Timing(msec): 500_

            Auto Guard? n    Call Still Held? n     Sig Bit Inversion: none
     Analog Loss Group: ___                             Digital Loss Group: ___
                                  Trunk Gain: high
              Bit Rate: 1200          Synchronization: _____    Duplex: ____
Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 10           Receive Answer Supervision? n
```

2. Type **1245** in the Night Service field.

   The destination can be a station extension, a recorded announcement extension, a vector directory number, a hunt group extension, a terminating extension group, or attd if you want to direct the call to the attendant.

3. Press ENTER to save your changes.

Setting up night service for hunt groups

You can administer hunt group night service if you want to direct hunt group calls to a night service destination.

Let's say your helpline on hunt group 3 does not answer calls after 6:00 p.m. When customers call after hours, you would like them to hear an announcement that asks them to try their call again in the morning.

Instructions

To set up night service for your helpline, you need to record the announcement (in our example, the announcement is on extension 1234) and then modify the hunt group to send calls to this extension.

To administer the hunt group for night service:

1. Type **change hunt-group 3** and press RETURN.

   The Hunt Group screen appears for hunt group 3.

```
                                    HUNT GROUP

          Group Number: 3                                             ACD? n
            Group Name: Accounting                                  Queue? y
       Group Extension: 2011                                       Vector? n
            Group Type: ucd-mia                             Coverage Path: 1
                    TN: 1                       Night Service Destination: 1234____
                   COR: 1                                 MM Early Answer? n
         Security Code: ____
      ISDN Caller Disp: ________

          Queue Length: 4
Calls Warning Threshold: ___   Port: x___   Extension: _______
 Time Warning Threshold: ___   Port: x___   Extension: _______
```

2. In the Night Service Destination field, type **1234**.

   The destination can be an extension, a recorded announcement extension, a vector directory number, a hunt group extension, or **attd** if you want to direct calls to the attendant.

   Calls to hunt group 3 will follow the coverage path assigned to extension 1234.

3. Press ENTER to save your changes.

4. Now you need to program a night service button.

   Refer to "Adding feature buttons" on page 127 for more information.

Related topics

"Managing hunt groups" on page 222.

How do night service types interact?

Let’s look at an example of how several types of night service might be used in one company.

Assume that you already administered the following night service settings:

- Night station night service redirects to extension 3000 and DID-LDN only to LDN Night Ext is set to **n**
- EXT Alert Port (TAAS) field is not defined
- Trunk group 4 redirects to extension 2000

Let’s look at how calls for this company are directed after hours:

call type	directs to
An LDN call on a DID trunk	extension 3000
A call on trunk group 4	extension 2000
An internal call to ‘0’	extension 3000
A call that redirects to the attendant through a coverage path	the attendant queue

Adding call pickup

To give your users the ability to pickup other users’ calls, you may want to use Call Pickup. To do this, you need to define a call pickup group.

Users may want to be able to pick up a call that is ringing at a nearby desk. Call Pickup provides 3 ways to pick up calls ringing at another phone:

- With Call Pickup, you create a call pickup group. All group members can answer a call ringing at another phone in the group from their own phone. If more than one phone is ringing, the one that has been ringing the longest is picked up.
- With Directed Call Pickup, users specify which ringing phone they want to answer from their own phone. A call pickup group is not required.
- With Group Call Pickup, users within an “extended” group can answer calls outside of their immediate group by entering a feature access code (FAC) followed by the 1- or 2-digit pickup (index) number.

Creating pickup groups

A pickup group is a list of phones where each member of the group can answer another member's calls. For example, if you want everyone in the payroll department to be able to answer calls to any payroll extension (in case someone is away from their desk), create a pickup group that contains all of the payroll extensions. Members of a pickup group should be located in the same area so that they can hear when the other extensions in the group ring.

Note that each extension may belong to only one pickup group. Also, the maximum number of pickup groups may be limited by your system configuration.

To create a pickup group:

1. Type **add pickup-group next** and press RETURN.

   The Pickup Group screen appears. The system selects the next Group Number for the new pickup group.

2. Enter the extension of each group member.

   Up to 50 extensions can belong to one group.

3. Press ENTER to save your new group list.

   The system automatically completes the name field when you press ENTER to save your changes.

```
                                   PICKUP GROUP

                    Group Number:          Extended Group Number: ____

GROUP MEMBER ASSIGNMENTS

   Ext      Name (first 26 characters)      Ext      Name (first 26 characters)
 1:                                         14:
 2:                                         15:
 3:                                         16:
 4:                                         17:
 5:                                         18:
 6:                                         19:
 7:                                         20:
 8:                                         21:
 9:                                         22:
10:                                         23:
11:                                         24:
12:                                         25:
13:
```

Once you define a pickup group, you can assign call-pickup buttons for each phone in the group or you can give each member the call-pickup feature-access code. Use the Station screen to assign call-pickup buttons.

To allow users to answer calls that are not in their pickup group, you may be able to use Directed Call Pickup.

Setting up directed call pickup

To set up a phone so that the user can pick up calls with Directed Call Pickup, you need to determine if directed call pickup is enabled on your system and make sure that the user's phone has a COR that allows directed call pickup.

To determine if Directed Call Pickup is enabled on your system:

1. Type **change system-parameters features** and press RETURN.

   The Feature-Related System Parameters screen appears.

2. Move to the Directed Call Pickup? field and enter **Y**.

3. Press ENTER to save the changes.

Now let's modify extension 4444 to allow directed call pickup. For this example, assume that the Can Use Directed Call Pickup field for COR 5 is set to Y.

1. Type **change station 4444** and press RETURN.

   The Station screen appears.

2. In the COR field, enter **5**.

3. Press ENTER to save your changes.

Setting up "simple" extended group pickup

Let's add a pickup group to an existing extended group where all members of the extended group pick up each other's calls. We will add pickup group 6 to the existing extended group 56.

To create a simple extended pickup group:

1. Type **change system-parameters features** and press RETURN.

   The Feature-Related System Parameters screen appears.

2. In the Extended Group Call Pickup field, type **simple**.

   Permits feature access codes to be administered.

3. Press ENTER to save your changes.

4. Type **change feature-access-codes** and press RETURN.

   The Feature Access Code screen appears.

5. In the Extended Group Call Pickup Access Code field, type the desired FAC.

   Refer to the dial plan to enter the correct sequence and number of digits.

6. Press ENTER to save your changes.

7. Type **add pickup-group next** and press RETURN.

   The Pickup Group screen appears.

```
                                PICKUP GROUP

                  Group Number: 1        Extended Group Number: ____

GROUP MEMBER ASSIGNMENTS

   Ext       Name (first 26 characters)          Ext      Name (first 26 characters)
 1:51001     station  51001                   14:51001     station  51002
 2:                                           15:
 3:                                           16:
 4:                                           17:
 5:                                           18:
 6:                                           19:
 7:                                           20:
 8:                                           21:
 9:                                           22:
10:                                           23:
11:                                           24:
12:                                           25:
13:
```

8. Enter the primary extensions of the users desired in the pickup group. Administer all the pickup groups that will be grouped together in the same extended pickup group.

9. Type **change extended-pickup-group n** and press RETURN, where n is the number of the extended pickup group to change.

   The Extended Pickup Group screen appears.

```
change extended-pickup-group 1                                  Page   1 of   2
                              EXTENDED PICKUP GROUP

                            Extended Group Number: 56

Pickup   Pickup Group                             Pickup   Pickup Group
Number     Number                                 Number     Number

 0:         345                                    13:        _____
 1:          25                                    14:        _____
 2:           2                                    15:        _____
 3:        _____                                   16:        _____
 4:        _____                                   17:        _____
 5:        _____                                   18:        _____
 6:        _____                                   19:        _____
 7:        _____                                   20:        _____
 8:        _____                                   21:        _____
 9:        _____                                   22:        _____
10:        _____                                   23:        _____
11:        _____                                   24:        _____
12:        _____                                   25:        _____
```

10. In the Pickup Group Number column, enter the pickup group number of the pickup groups that belong to the extended group being administered.

    The Pickup Number associated with the entered Pickup Group Number is the number users will enter following the FAC when pickup group. In the above listed example, a user in Pickup Group 2 will enter “0” following the FAC to pick up a call from pickup group 345.

    To minimize the number of digits entered by users, the Pickup Numbers should be the lowest available Pickup Numbers.

11. Press ENTER to save your changes.

Setting up “flexible” extended group pickup

To create a flexible extended pickup group:

1. Type **change system-parameters features** and press RETURN.

   The Feature-Related System Parameters screen appears.

2. In the Extended Group Call Pickup field, type **flexible**.
3. Press ENTER to save your changes.
4. Type **change feature-access-codes** and press RETURN.

   The Feature Access Code screen appears.

5. In the Extended Group Call Pickup Access Code field, type the desired FAC.

   Refer to the dial plan to enter the correct sequence and number of digits.

6. Press ENTER to save your changes.
7. Type **add pickup-group next** and press RETURN.

   The Pickup Group screen appears.

8. In the Extended Group Number field, type the number of the extended pickup group that can answer calls to this group.

   For each administered pickup group, indicate an extended group by entering the extended pickup group number. This extended pickup group number identifies the other pickup groups that can have calls picked up by the pickup group.

9. Add extended members as for other pickup features.

10. Type **change extended-pickup-group n** and press RETURN, where n is the number of the extended pickup group to change.

    The Extended Pickup Group screen appears.

```
change extended-pickup-group 1                                 Page   1 of   2
                              EXTENDED PICKUP GROUP

                            Extended Group Number: 56

Pickup   Pickup Group                              Pickup   Pickup Group
Number     Number                                  Number     Number

 0:         345                                     13:       _____
 1:          25                                     14:       _____
 2:          2                                      15:       _____
 3:       _____                                     16:       _____
 4:       _____                                     17:       _____
 5:       _____                                     18:       _____
 6:       _____                                     19:       _____
 7:       _____                                     20:       _____
 8:       _____                                     21:       _____
 9:       _____                                     22:       _____
10:       _____                                     23:       _____
11:       _____                                     24:       _____
12:       _____                                     25:       _____
13:
```

11. In the Pickup Group Number column, type the pickup group numbers of the pickup group that can answer each others calls.

    This screen groups together the pickup groups that can have their calls picked up by the pickup groups with identical extended pickup group numbers. The pickup groups numbers are listed using the lowest available Pickup Numbers. A unique “Extended Pickup Group number can be assigned to each pickup group on this screen. The entries in the Extended Group are the groups that can have calls picked up by the group with the matching Extended Group number in the Pick Group screen. Thus, a particular group’s calls may be picked up by members of many extended pickup groups.

    NOTE:
    On the Feature-Related System Parameters screen, if the Extended Group Call Pickup field is **flexible**, you can administer a pickup group in multiple extended pickup groups.

12. Press ENTER to save your changes.

Managing hunt groups

This section shows you how to set up hunt groups. It explains how calls to a hunt group are handled and shows you different call distribution methods.

What are hunt groups?

A hunt group is a group of extensions that receive calls according to the call distribution method you choose. When a call is made to a certain phone number, the system connects the call to an extension in the group.

Use hunt groups when you want more than one person to be able to answer calls to the same number. For example, set up a hunt group for:

- a benefits department within your company
- a travel reservations service

Setting up hunt groups

Let’s set up a hunt group for an internal helpline. Before making changes to the switch, we’ll decide:

- the phone number for the hunt group
- the number of people answering calls
- the way calls are answered

Our dial plan allows 4-digit internal numbers that begin with 1. The number 1200 is not in use. So, we’ll set up a helpline hunt group so anyone within the company can call extension 1200 for help with a phone.

We will assign 3 people (agents) and their extensions to our helpline. We want calls to go to the first available person.

Instructions

To set up our helpline hunt group:

1. Type **add hunt-group next** and press RETURN.

   The Hunt Group screen appears. The Group Number field is automatically filled in with the next hunt group number.

```
                                                               Page 1 of X
                                       HUNT GROUP

           Group Number: 4__                                      ACD? _
             Group Name: internal helpline____________           Queue? _
        Group Extension: 1200____                               Vector? _
             Group Type: ucd-loa                         Coverage Path: ____
                     TN: ___                 Night Service Destination: _____
                    COR: _                             MM Early Answer? _
     Security Code: ____
  ISDN Caller Disp: ________
```

2. In the Group Name field, type the name of the group.

   In our example, type **internal helpline**.

3. In the Group Extension field, type the phone number.

   We'll type **1200**.

4. In the Group Type field, type the code for the call distribution method you choose.

   We'll type **ucd-loa** so a call goes to the agent with the lowest percentage of work time since login.

   NOTE:
   The COS for all hunt groups defaults to 1. Therefore, any changes to COS 1 on the Class of Service screen changes the COS for all your hunt groups. A COS field does not appear on the Hunt Group screen.

5. Press NEXT PAGE to find the Group Member Assignments page.

```
 HUNT GROUP
          Group Number: 4 Group Extension: 3001        Group Type: ucd
  Member Range Allowed: 1 - 999         Administered Members (min/max): 1  /9
                                            Total Administered Members: 9
GROUP MEMBER ASSIGNMENTS
      Ext       Name (24 characters)             Ext       Name (24 characters)

 1 :1011                                    14 :_______
 2 :1012                                    15 :_______
 3 :1013                                    16 :_______
 4 :_________                               17 :_______
 5 :_________                               18 :_______
 6 :_________                               19 :_______
 7 :________                                20 :_______
 8 :________                                21 :_______
 9 :________                                22 :_______
10 :________                                23 :_______
11 :________                                24 :_______
12 :________                                25 :_______
13 :________                                26 :_______

More Members Exist
```

6. In the Ext field, type the extensions of the agents you want in the hunt group.

   We'll type **1011**, **1012**, and **1013**.

   Tip:
   For a ddc group type (also known as "hot seat" selection), the call is sent to the extension listed in the first Ext field. The system uses this screen to determine the hunting sequence.

7. Press ENTER to save your changes.

   The Name fields are display-only and do not appear until the next time you access this hunt group.

Related topics

Refer to ''Hunt Group'' on page 840 for more information on an ACD and non-ACD hunt group.

Changing a hunt group

To make changes to a hunt group:

1. Type **change hunt-group n** and press RETURN, where n is the number of the hunt group.
2. Change the necessary fields.
3. Press ENTER to save your changes.

Setting up a queue

You can tell your switch how to handle a hunt-group call when it cannot be answered right away. The call waits in a “queue.”

Let’s tell the switch that up to 10 calls can wait in the queue, but that you want to be notified if a call waits for more than 30 seconds.

You also want the switch to send a warning when 5 or more calls are waiting in the queue. This warning flashes queue-status buttons on phones that have a status button for this hunt group. When the buttons flash, everyone answering these calls can see that the help-line calls need more attention.

Instructions

To set up our helpline queue:

1. Type **change hunt-group n** and press RETURN, where n is the number of the hunt group to change.

   In our example, type **change hunt-group 5**. The Hunt Group screen appears.

```
                                                                   Page 1 of X
                                    HUNT GROUP

              Group Number: 4__                                      ACD? _
                Group Name: internal helpline____________          Queue? _
           Group Extension: 1200____                              Vector? _
                Group Type: ucd-loa                         Coverage Path: ____
                        TN: ___                  Night Service Destination: _____
                       COR: _                               MM Early Answer? _
        Security Code: ____
     ISDN Caller Disp: ________


                    Measured: ________            Supervisor: Extension: ____

         Controlling Adjunct: ____
      Multiple Call Handling: __________
                   Objective: ____

              Queue Length: 10
   Calls Warning Threshold: 5                 Calls Warning Port: __
```

2. In the Queue field, type **y**.
3. In the Queue Length field, type the maximum number of calls that you want to wait in the queue.

   In our example, type **10**.

4. In the Calls Waiting Threshold field, type the maximum number of calls that can be in the queue before the system flashes the queue status buttons.

   In our example, type **5**.

5. In the Time Warning Threshold field, type the maximum number of seconds you want a call to wait in the queue before the system flashes the queue status buttons.

   In our example, type **30**.

6. Press ENTER to save your changes.

Adding hunt group announcements

You can add recorded announcements to a hunt group queue. Use announcements to encourage callers to stay on the line or to provide callers with information. You can define how long a call remains in the queue before the caller hears an announcement.

Refer to ‘‘Recording announcements’’ on page 434 for information on how to record an announcement.

Let’s add an announcement to our internal helpline. We want the caller to hear an announcement after 20 seconds in the queue, or after approximately 4 or 5 rings. Our announcement is already recorded and assigned to extension 1234.

Tip:
You can use ***display announcements*** *to find the extensions of your recorded announcements.*

Instructions

To add an announcement to our helpline queue:

1. Type **change hunt-group n** and press RETURN, where n is the number of the hunt group to change.

   In our example, type **change hunt-group 5**.

   The Hunt Group screen appears.

2. Press NEXT PAGE to find the First Announcement Extension field.

```
                              HUNT GROUP

                       Message Center: ________
                      AUDIX Extension: ________
            Message Center AUDIX Name: _______
                             Primary? _
  Calling Party Number to INTUITY AUDIX? _
                        LWC Reception: _______
                           AUDIX Name: _______
                  Messaging Server Name: _______

         First Announcement Extension: 1234 ___ Delay (sec): 20
        Second Announcement Extension: ________ Delay (sec): __ Recurring? _
```

3. In the First Announcement Extension field, type the extension of the announcement you want callers to hear.

   In our example, type **1234**.

4. In the First Announcement Delay (sec) field, type the number of seconds you want the caller to wait before hearing the first announcement.

   In our example, type **20**.

   Tip:
   If you set the delay announcement interval to 0, callers automatically hear the announcement before anything else. This is called a "forced first announcement."

5. Press ENTER to save your changes.

You can use the same announcement for more than one hunt group.

Managing vectors and VDNs

This section provides an introduction to vectors and Vector Directory Numbers (VDN). It gives you basic instructions for writing simple vectors.

SECURITY ALERT:

*Vector fraud is one of the most common types of toll fraud because vectors route calls based on the Class of Restriction (COR) assigned to the VDN. R*efer to *Avaya Products Security Handbook for more information.*

This section references announcements, hunt groups, queues, splits, and skills, which are covered in detail in other sections of this book. You can also find information about these topics in the Avaya *Call Vectoring/Expert Agent Selection (EAS) Guide*.

What are vectors?

A vector is a series of commands that you design to tell the system how to handle incoming calls. A vector can contain up to 32 steps and allows customized and personalized call routing and treatment. Use call vectoring to:

- play multiple announcements
- route calls to internal and external destinations
- collect and respond to dialed information

Tip:

The vector follows the commands in each step in order. The vector "reads" the step and follows the command if the conditions are correct. If the command cannot be followed, the vector skips the step and reads the next step.

Your system can handle calls based on a number of conditions, including the number of calls in a queue, how long a call has been waiting, the time of day, day of the week, and changes in call traffic or staffing conditions.

Writing vectors

Writing vectors is easy, but we recommend that you set up and test your vectors before you use them across the system.

We'll write a vector to handle calls to our main number. It is the first vector so we'll use number 1.

Tip:

Use **list vector** *to see a list of existing vectors.*

Before you start

- On the System-Parameters Customer-Options screen, verify the Basic Call Vectoring field is **y**. If not, contact your Avaya representative.
- To provide announcements, you need an Announcement circuit pack. Refer to Avaya MultiVantage™ Hardware Solutions Guide for more information on the circuit pack.
- Use one of the following:
  - Tone Clock with Call Classifier - Tone Detector circuit pack.
  - Call Classifier - Detector circuit pack.

Instructions

To write a vector:

1. Type **change vector 1** and press RETURN.

   The Call Vector screen appears.

```
                                  CALL VECTOR
    Number:  1        Name: main number   calls ______ Multimedia? n      Lock? n


     Basic? y    EAS? n     G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
 Prompting? y    LAI? n    G3V4 Adv Route? n           CINFO? n             BSR? n

   01 ____________
   02 ____________
   03 ____________
   04 ____________
   05 ____________
   06 ____________
   07 ____________
   08 ____________
   09 ____________
   10 ____________
   11 ____________
```

   The vector Number field on the left side of the screen is filled in automatically.

2. In the Name field, type a description for the vector.

   In our example, type **main number calls**.

   Tip:
   The information in the heading of the Call Vector screen is display only. Use ***display system-parameters customer-options*** *to see the features that are turned on in your switch.*

3. Type your vector steps in the numbered column on the left of the screen.

> **Tip:**
> *When you type in your vector steps, the switch automatically completes some of the vector step information for you. For example, if you type "q" in a vector step field, the switch fills in "queue-to." Also, additional fields appear when you complete a field and press* TAB*. This makes it very easy to type in your vector steps.*

Now that vector 1 is set up, let's add a vector step to it to tell the switch how to handle the calls to our main number.

Putting a call in a queue

Write a vector so that calls that come into the main business number redirect to a queue.

We'll use a vector-controlled hunt group for the main number queue. This hunt group was set up as main split 47. When calls first arrive, all calls to our main number should be queued as "pri 1" for low priority.

To queue calls, write the following vector (step 2). (Please note, we started our example on step 2 because step 1 is used later in this chapter.)

```
                              CALL VECTOR
   Number:  1       Name: main number   calls ______ Multimedia? n     Lock? n


    Basic? y   EAS? n    G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
Prompting? y   LAI? n   G3V4 Adv Route? n          CINFO? n             BSR? n

  01 ____________
  02 queue-to main split 47 pri 1
  03 ____________
  04 ____________
  05 ____________
  06 ____________
  07 ____________
  08 ____________
  09 ____________
  10 ____________
  11 ____________
```

> **Tip:**
> *Remember, the switch automatically fills in some of the information when you type your vector step and press* TAB*.*

Playing an announcement

Write a vector to play an announcement for callers in a queue. Use the announcement to ask callers to wait. You need to record the announcement before the vector can use it.

Let's play our announcement 4001, asking the caller to wait, then play music for 60 seconds, then repeat the announcement and music until the call is answered. The **goto** command creates the loop to repeat the announcement and the music. **Unconditionally** means under all conditions.

Tip:

Rather than loop your vectors directly back to the announcement step, go to the previous queue-to step. This way, if for some reason the call does not queue the first time, the switch can attempt to queue the call again. If the call successfully queued the first time though, it merely skips the queue-to step and plays the announcement. The system cannot queue a call more than once in the exact same priority level.

To play and repeat an announcement, write this vector (steps 3-5):

```
                                CALL VECTOR
   Number:  1       Name: main number   calls ______ Multimedia? n     Lock? n


    Basic? y   EAS? n    G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
Prompting? y   LAI? n   G3V4 Adv Route? n           CINFO? n            BSR? n

  01 ____________
  02 queue-to main split 47 pri 1
  03 announcement 4001 (All agents are busy, please wait...)
  04 wait-time 60 secs hearing music
  05 goto step 2 if unconditionally
  06 ____________
  07 ____________
  08 ____________
  09 ____________
  10 ____________
  11 ____________
```

Routing based on time of day

Write a vector for calls that come in after your office closes.

Assume that your business is open 7 days a week, from 8:00 a.m. to 5:00 p.m. When calls come in after business hours, you want to play your announcement 4002, which states that the office is closed and asks callers to call back during normal hours. Write the vector so the call disconnects after the announcement is played.

For after hours treatment, write this vector (steps 1, 6, and 7):

```
                              CALL VECTOR
   Number:  1       Name: main number   calls ______ Multimedia? n     Lock? n


    Basic? y   EAS? n    G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
Prompting? y   LAI? n   G3V4 Adv Route? n           CINFO? n            BSR? n

  01 goto step 7 if time-of-day is all 17:00 to all 8:00
  02 queue-to main split 47 pri 1
  03 announcement 4001 (All agents are busy, please wait...)
  04 wait-time 60 secs hearing music
  05 goto step 2 if unconditionally
  06 stop
  07 disconnect after announcement 4002 (“We’re sorry, our office is closed...“)
  08 ____________
  09 ____________
  10 ____________
  11 ____________
```

If the **goto** command in step 5 fails, the switch goes to the next step. The **stop** in step 6 prevents callers from incorrectly hearing the “office is closed” announcement in step 7. **Stop** keeps the call in the state it was in before the command failed. In this case, if step 5 fails, the call remains in step 4 and the caller continues to hear music.

CAUTION:
Add a stop vector step only after calls are routed to a queue. If a stop vector is executed for a call not in queue, the call drops.

Allowing callers to leave a message

Write a vector that allows callers to leave messages. This type of vector uses a hunt group called a messaging split. For our example, we send after-hours calls to the voice mailbox at extension 2000 and use messaging split 99.

Once the vector routes a call to the mailbox, the caller hears a greeting (that was recorded with the voice mail for mailbox 2000) that tells them they can leave a message.

To let callers leave messages, write this vector (step 7):

```
                              CALL VECTOR
   Number:  1       Name: main number   calls ______ Multimedia? n     Lock? n


    Basic? y    EAS? n     G3V4 Enhanced? n    ANI/II-Digits? n    ASAI Routing? n
Prompting? y    LAI? n    G3V4 Adv Route? n            CINFO? n             BSR? n

  01 goto step 7 if time-of-day is all 17:00 to all 8:00
  02 queue-to main split 47 pri 1
  03 announcement 4001 (All agents are busy, please wait...)
  04 wait-time 60 secs hearing music
  05 goto step 2 if unconditionally
  06 stop
  07 messaging split 99 for extension 2000
  08 ____________
  09 ____________
  10 ____________
  11 ____________
```

Redirecting calls during an emergency or holiday

You can provide a quick way for a supervisor or agent to redirect calls during an emergency or holiday. Use a special mailbox where you can easily change announcements. This vector is also an alternative to making sure all agents log out before leaving their phones.

In our example, no agents are normally logged in to split 10. We'll use split 10 for an emergency. We preset buttons on our agents' phones so people with these phones can log in at the touch of a button.

To quickly redirect calls:

1. Create a special mailbox with the appropriate announcement such as "We are unable to answer your call at this time" or "Today is a holiday, please call back tomorrow."

   In our example, we recorded the mailbox greeting for extension 2001.

2. Insert the following bold steps (steps 1, 10, and 11).

   Refer to "Inserting a step" on page 235 for more information.

```
                              CALL VECTOR
 Number:  1       Name: main number   calls ______ Multimedia? n     Lock? n


  Basic? y   EAS? n    G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
Prompting? y   LAI? n   G3V4 Adv Route? n          CINFO? n            BSR? n

01 goto step 10 if staff agents split 10 > 0
02 goto step 8 if time-of-day is all 17:00 to all 8:00
03 queue-to main split 47 pri 1
04 announcement 4001 (All agents are busy, please wait...)
05 wait-time 60 secs hearing music
06 goto step 2 if unconditionally
07 stop
08 messaging split 99 for extension 2000
09 stop
10 messaging split 99 for extension 2001
11 stop
```

When there is an emergency, fire drill, or holiday, the supervisor or agent logs into this split.

When an agent logs into split 10, the system looks at vector step 1, sees that more than 0 people are logged into split 10, and sends calls to step 10 (which sends to messaging split 99).

When your business returns to normal and the agent logs out of split 10, call handling returns to normal.

Giving callers additional choices

You can give your callers a list of options when they call. Your vector tells the switch to play an announcement that contains the choices. The switch collects the digits the caller dials in response to the announcement and routes the call accordingly.

We’ll create a vector that plays an announcement, then lets callers dial an extension or wait in the queue for an attendant.

Please note, the following example of this “auto attendant” vector is a new vector and is not built on the vector we used in the previous example.

To let callers connect to an extension, write this kind of vector:

```
                              CALL VECTOR
  Number:  1       Name: main number   calls ______ Multimedia? n     Lock? n


   Basic? y   EAS? n    G3V4 Enhanced? n   ANI/II-Digits? n   ASAI Routing? n
Prompting? y   LAI? n   G3V4 Adv Route? n           CINFO? n               BSR? n

  01 wait-time 0 seconds hearing music
  02 collect 4 digits after announcement 4004 (You have reached our company.
     Please dial a 4-digit extension or wait for the attendant.)
  03 route-to digits with coverage y
  04 route-to number 0 with cov n if unconditionally
  05 stop
  06 ____________
  07 ____________
  08 ____________
  09 ____________
  10 ____________
```

Inserting a step

It is easy to change a vector step and not have to retype the entire vector. Let's add announcement 4005 between step 3 and step 4 in vector 20.

To insert a new vector step in vector 20:

1. Type **change vector 20** and press RETURN.

   The Call Vector screen appears.

2. Press EDIT.
3. Type **i** followed by a space and the number of the step you want to add.

   In our example, type **i 4**.

4. Type the new vector step.

   We'll type **announcement 4005 (Please wait...)**.

5. Press ENTER to save your changes.

Tip:
When you insert a new vector step, the system automatically renumbers the rest of the vector steps and all references to the vector steps. The switch inserts a "" when the numbering needs more attention.*

Deleting a step

To delete vector step 5 from vector 20:

1. Type **change vector 20** and press RETURN.

   The Call Vector screen appears.

2. Press EDIT.

3. Type **d** followed by a space and the number of the step you want to delete.

   In our example, type **d 5**.

   Tip:
   *You can delete a range of vector steps. For example, to delete steps 2 through 5, type **d 2-5** and press* ENTER.

4. Press ENTER to save your changes.

   Tip:
   When you delete a vector step, the system automatically renumbers the rest of the vector steps and all references to the vector steps. The switch inserts a "" when the numbering needs more attention.*

More information

Refer to Avaya *Call Vectoring/Expert Agent Selection (EAS) Guide* for more information.

Automated Attendant competes with several features for ports on the Call Classifier — Detector circuit pack or equivalent. Refer to Avaya MultiVantage™ Hardware Solutions Guide for more information on the circuit pack.

Fixing problems

If there is a problem with a vector, the switch records the error as a vector event. Vector events occur for a number of reasons including problems with a trunk, full queue slots, or the vector reaching the maximum 1000 steps allowed.

Use **display events** to access the Event Report screen and see the event record. Use the event record to see why the vector failed.

To view the Event Report:

1. Type **display events** and press RETURN.

   The Event Report screen appears.

```
                              EVENT REPORT
The following option control which events will be displayed:
   EVENT CATEGORY
               Category: Vector
   REPORT PERIOD
               Interval: _a_ From: __/__/__:__ To: __/__/__:__
   SEARCH OPTIONS
               Vector Number: __
               Event Type: ___
```

2. To see all current vector events, press RETURN.

   OR

   Indicate the events that you want to see by completing the Report Period and Search Option fields. Refer to *Avaya MultiVantage™ Call Center Software Release 11Call Vectoring and Expert Agent Selection (EAS) Guide* for more information.

3. Press ENTER to view the report.

   The Event Report (detail) screen appears.

```
                                       EVENT REPORT
Event Event                     Event    Event    First         Last          Event
Type  Description                Data 1   Data 2   Occur         Occur         Cnt
  20  Call not queued            12/5     B        09/28/13:43  09/28/13:43   21
 541  Not a messaging split      Split    4C       09/28/13:43  09/28/13:43   136
```

   Look at the information in Event Data field to diagnose the vector event. In this example, there was a problem with:

   - Vector 12, step 5
   - Split 89

Vector directory numbers

A vector directory number (VDN) is an extension that directs an incoming call to a specific vector. This number is a “soft” extension number not assigned to an equipment location. VDNs must follow your dial plan.

Let's create VDN 5011 for our sales department. A call into 5011 routes to vector 11. This vector plays an announcement and queues calls to the sales department.

SECURITY ALERT:

Vector fraud is one of the most common types of toll fraud because vectors route calls based on the class of restriction (COR) assigned to the VDN. Refer to Avaya Products Security Handbook for more information.

Adding a vector directory number

To add a vector directory number:

1. Type **add VDN 5011** and press RETURN.

   You enter the VDN extension you want to add. The Vector Directory Number screen appears.

```
                           VECTOR DIRECTORY NUMBER

                            Extension: 5011
                                 Name: Sales Department
                  Allow VDN Override? n
                                  COR: 1
                                   TN: 1
                       Vector Number: 11
                          AUDIX Name:
               Messaging Server Name:
                            Measured: both
  Acceptable Service Level (sec):
   VDN of Origin Annc. Extension: 301
                           1st Skill:
                           2nd Skill:
                           3rd Skill:

                  Return Destination:
              VDN Timed ACW Interval:
                     BSR Application:
    BSR Available Agent Strategy: 1st-found
```

2. Type a description for this VDN in the Name field.

   In our example, type **Sales Department**.

   The information in the VDN Name field appears on a display phone. This allows the agent to recognize the nature of the call and respond accordingly.

   Tip:

   The VDN Override on the Vector Directory Number screen controls the operation of the display.

3. Enter the vector number.

   In our example, type **11**.

4. In the Measured field, indicate how you want to measure calls to his VDN.

   In our example, type **both** (for both CMS and BCMS).

   Tip:
   BCMS must be enabled to use "both." Use ***display system-parameters customer-options*** *to see if BCMS is enabled.*

5. Press ENTER to save your changes.

Viewing vector directory numbers

To see the VDNs already associated with your vectors:

1. Type **list VDN** and press RETURN.

   The Vector Directory Number screen appears.

```
                          VECTOR DIRECTORY NUMBER
                                                                    Event
                                   VDN          Vec         Orig    Notif  Skills
Name                          Ext   Ovrd  COR TN Num  Meas  Annc    Adj 1st 2nd 3rd
Tech Support                  5000  y     59  1  234  none  301
Customer Serv.                5001  n     1   1    1  none  302
New Orders                    5002  y     23  1    5  none  303
```

Each VDN maps to one vector. Several VDNs can map to the same vector.

Understanding Automatic Call Distribution

Automatic Call Distribution (ACD) is MultiVantage feature used in many call centers. ACD gives you greater flexibility to control call flow and to measure the performance of agents.

ACD systems operate differently from non-ACD systems, and they can be much more complex. ACD systems can also be more powerful because they allow you to use features and products that are not available in non-ACD systems. Refer to *Avaya MultiVantage™ Call Center Software Release 11 Guide to ACD Call Centers* for more information on ACD call centers.

Enhancing an ACD system

First, all call center management systems (such as Avaya's Basic Call Management System (BCMS), BCMSVu, and the sophisticated Avaya IP Agent Call Management System) require ACD. These management systems give you the ability to measure more aspects of your center's operation, and in more detail, than is possible with standard Avaya MultiVantage reports.

Call vectoring greatly enhances the flexibility of a call center, and most vectoring functions require ACD. Vectoring is a simple programming language that allows you to custom design every aspect of call processing. Refer to ''What are vectors?'' on page 228 for more information.

Together, ACD and vectoring allow you to use Expert Agent Selection (EAS). For a variety of reasons, you may want certain agents to handle specific types of calls. For example, you may want only your most experienced agents to handle your most important customers. You may have multilingual agents who can serve callers in a variety of languages.

EAS allows you to classify agents according to their specific skills and then to rank them by ability or experience within each skill. Avaya MultiVantage uses these classifications to match each call with the best available agent. Refer to *Avaya MultiVantage™ Call Center Software Release 11Call Vectoring and Expert Agent Selection (EAS) Guide* for more information on call vectoring and EAS.

Assigning a terminating extension group

A Terminating Extension Group (TEG) allows an incoming call to ring as many as 4 phones at one time. Any user in the group can answer the call.

Once a member of the TEG has answered a group call, the TEG is considered busy. If a second call is directed to the group, it follows a coverage path if one has been assigned.

Instructions

Now assign a terminating extension group to the advertising department. For example, let's assign this TEG to extension 6725.

1. Type **add term-ext-group next** and press RETURN.

   The Terminating Extension Group screen appears.

```
                         TERMINATING EXTENSION GROUP

    Group Number: 1                                 Group Extension: 6725
      Group Name: advertising                         Coverage Path: 5
   Security Code:                                               COR: 1
                                                                 TN: 1
ISDN Caller Disp: mbr-name                            LWC Reception: none
      AUDIX Name:                               Messaging Server Name:

GROUP MEMBER ASSIGNMENTS

    Ext     Name (first 26 characters)      Ext     Name (first 26 characters)
 1: 5101    26 character name sta 5101   3:
 2:                                      4: 5102    26 character name sta 5102
```

2. In the Group Extension field, type **6725**.

   This is the extension for the advertising group.

3. In the Group Name, type **advertising**.

   This is the name of the group.

4. In the Coverage Path field, type **5**.

   This is the number of the call coverage path for this group.

5. In the COR field, leave the default as **1**.
6. In the TN field, leave the default as **1**.
7. In the ISDN Call Display field, type **mbr-name**.

   This specifies that the member name (member of the TEG where the call terminated) is sent to the originating user.

8. In the Ext field, in the 1st place, type **5101**.
9. In the 4th place, type **5102**.
10. Press ENTER to save your changes.

11. Type **change station 6725** and press RETURN.

    The Station screen for extension 6725 appears.

```
                                                              Page 2 of X
                                          STATION
FEATURE OPTIONS
           LWC Reception? msa-spe         Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
             CDR Privacy? n                                   Auto Answer: none
   Redirect Notification? y                              Data Restriction? n
 Per Button Ring Control? n                    Idle Appearance Preference? n
  PCOL/TEG Call Alerting? n
  Active Station Ringing: single                    Restrict Last Appearance? y
                                     Per Station CPN - Send Calling Number? _
        H.320 Conversion? n
              AUDIX Name: ______
   Messaging Server Name: ______                     Audible Message Waiting? n
                                                 Display Client Redirection? n
                                                Select Last Used Appearance? n
```

12. In the Bridged Call Alerting field, type **y**.

    This provides audible ringing for TEG calls.

13. In the Button Assignments section, type **term-x-gr 1**.

    This is the TEG button for the advertising group.

14. Press ENTER to save your changes.

Routing outgoing calls

8

World class routing

Your system uses Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) to direct outgoing calls.

- AAR routes calls within your company over your own private network.
- ARS routes calls that go outside your company over public networks. ARS also routes calls to remote company locations if you do not have a private network.

Automatic routing begins when a user dials a Feature Access Code (FAC) followed by the number the user wants to call. The switch analyzes the digits dialed, selects the route for the call, deletes and inserts digits if necessary, and routes the call over the trunks you specify in your routing tables. ARS and AAR can access the same trunk groups and share the same route patterns and other routing information. ARS calls can be converted to AAR calls and vice-versa.

The FAC for AAR is usually the digit 8. The FAC for ARS is usually the digit 9 in the US and 0 outside of the US. Your Avaya technician sets up AAR on your switch and usually assigns the AAR FAC at the same time. You can administer your own ARS FAC.

This section describes only ARS call routing.

Managing calling privileges

Each time you set up a phone, you use the station screen to assign a COR. You can create different CORs for different groups of users. For example, you may want executives in your company to have different calling privileges than receptionists.

When you set up a COR, you specify a Facility Restriction Level (FRL) on the Class of Restriction screen. The FRL determines the calling privileges of the user. Facility Restriction Levels are ranked from 0–7, where 7 has the highest level of privileges.

You also assign an FRL to each route pattern preference in the route pattern screen. When a user makes a call, the system checks the user’s COR. The call is allowed if the caller’s FRL is higher than or equal to the route pattern preference’s FRL.

Instructions

Let’s say we are setting up a new phone for an executive. The current translations assign COR 1, with outward restrictions and an FRL 0, which is the lowest permission level available. We want to assign a COR with the highest level of permissions, FRL 7, to station 1234.

To change station 1234 from COR 1 to COR 7:

1. Type **change station 1234** and press RETURN.

   The Station screen appears.
2. In the COR field, type **7** and press ENTER to save your changes.
3. To change from FRL 0 to FRL 7, type **change cor 7** and press RETURN.

   The Class of Restriction screen appears.
4. In the FRL field, type **7** and press ENTER to save your changes.

   Now all users with COR 7 will have the highest level of calling permissions.

Assigning ARS FAC

Be sure the ARS Feature Access Code (FAC) is set up on your system. In the U.S., 9 is usually the ARS FAC. Users dial 9 to make an outgoing call.

When a user dials 9 to access ARS and make an outgoing call, the ARS access code 9 is dropped before digit analysis takes place. will not be part of the digit analysis.

Instructions

To assign the ARS FAC:

1. Type **change dialplan** and press ENTER.

   The DCS to QSIG TSC Gateway screen screen appears.

2. Move to the 9 row and type **fac** in the first column. Press ENTER to save your changes.

3. Type **change features** and press ENTER.

   The Feature Access Code screen appears.

4. Type **9** in the ARS - access code field and press ENTER to save your changes.

Displaying ARS analysis information

Instructions

You'll want to become familiar with how your system currently routes outgoing calls. To display the ARS Digit Analysis Table that controls how the system routes calls that begin with 1:

1. Type **display ars analysis 1** and press RETURN.

   The ARS Digit Analysis Table for dialed strings that begin with 1 appears. Note that the switch displays only as many dialed strings as can fit on one screen at a time.

To see all the dialed strings that are defined for your system, run an ARS Digit Analysis report:

1. Type **list ars analysis** and press RETURN.

   The ARS Digit Analysis Report appears. You may want to print this report to keep in your paper records.

Understanding ARS analysis

With ARS, the switch checks the digits in the number called against the ARS Digit Analysis Table to determine how to handle the dialed digits. Your switch also uses Class of Restriction (COR) and Facility Restriction Level (FRL) to determine the calling privileges.

Let's look at a very simple AAR and ARS Digit Analysis Table. Your system likely has more defined dialed strings than our example.

ARS DIGIT ANALYSIS TABLE

Location: all   Percent Full: 6

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req
1	1	1	12	svcl	___	n
1	11	11	30	fnpa	___	n
1	12	23	17	intl	___	n
10xxx	5	5	deny	op	___	n
1800	11	11	30	fnpa	___	n
2	7	7	2	hnpa	___	n
3	7	7	2	hnpa	___	n
4	7	7	2	hnpa	___	n
5	7	7	2	hnpa	___	n
6	7	7	2	hnpa	___	n
7	7	7	2	hnpa	___	n

The far-left column of the ARS Digit Analysis Table lists the first digits in the dialed string. When a user makes an outgoing call, the system analyzes the digits, looks for a match in the table, and uses the information in the matching row to determine how to route the call.

Let's say a caller places a call to 1-303-233-1000. The switch matches the dialed digits with those in the first column of the table. In this example, the dialed string matches the '1'. Then the systems matches the length of the entire dialed string (11 digits) to the minimum and maximum length columns. In our example, the 11-digit call that started with 1 follows route pattern 30 as an fnpa call.

Tip:

The first dialed digit for an external call is often an access code. If '9' is defined as the ARS access code, the switch drops this digit and analyzes the remaining digits with the ARS Analysis Table.

The Route Pattern points to the route that handles the calls that match this dial string.

Call Type tells what kind of call is made with this dial string. Call type helps the switch decide how to handle the dialed string.

Examples of digit conversion

Your system uses the AAR or ARS Digit Conversion Table to change a dialed number for more efficient routing. Digits may be inserted or deleted from the dialed number. For instance, you can tell the switch to delete a 1 and an area code on calls to one of your locations, and avoid long-distance charges by routing the call over your private network.

The table below reflects these values:

- ARS feature access code = 9
  AAR feature access code = 8
  Private Network Office Code (also known as Home RNX) = 222
  Prefix 1 is required on all long-distance DDD calls
  Dashes (-) are for readability only

The switch maps the dialed digits to the matching pattern that most closely matches the dialed number. Example: If the dialed string is 957-1234 and matching patterns 957-1 and 957-123 are in the table, the match is on pattern 957-123.

Table 4. ARS Digit Conversion Examples

Operation	Actual Digits Dialed	Matching Pattern	Replacement String	Modified Address	Notes
DDD call to ETN	9-1-303-538-1345	1-303-538	362	362-1345	Call routes via AAR for RNX 362
Long-distance call to specified carrier	9-10222+DDD	10222	(blank)	(blank)	Call routes as dialed with DDD # over private network
Terminating a local DDD call to an internal station	9-1-201-957-5567 or 9-957-5567	1-201-957-5 or 957-5	222-5	222-5567.	Call goes to home RNX 222, ext. 5567
Unauthorized call to intercept treatment	9-1-212-976-1616	1-XXX-976	#	(blank)	"#" means end of dialing. ARS ignores digits dialed after 976. User gets intercept treatment.
International calls to an attendant	9-011-91-672530	011-91	222-0111#	222-0111	Call routes to local switch (RNX 222), then to attendant (222-0111).
International call to announcement (This method may also be used to block unauthorized IDDD calls)	9-011-91-672530	011-91	222-1234#	222.1234-	Call routes to local switch (RNX 222), then to announcement extension (222-1234).
International call from certain European countries needing dial tone detection	0-00-XXXXXXXX	00	+00+	00+XXXX	The first 0 denotes ARS, the second pair of 0s denotes an international call, the pluses denote "wait" for dial tone detection.

Defining operator assisted calls

Let’s look at how the switch routes an ARS call that begins with 0 and requires operator assistance. Remember, the user dials 9 to access ARS, then a 0, then the rest of the number.

To see how your switch handles a call to an operator:

1. Type **display ars analysis 0** and press ENTER.

   The AAR and ARS Digit Analysis Table screen starting with 0 appears.

```
                         ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6
         Dialed           Total      Route    Call  Node  ANI
         String          Min Max     Pattern Type   Num   Req
   0____________________1    1       1       svcl   ___   n
   0____________________8__  8 ____  1       op     ___   n
   0____________________11_  11____  1       op     ___   n
   00___________________2 _  2 ____  1       op     ___   n
   01___________________10_  23 ___  deny    op     ___   n
   011__________________10_  23 ___  deny    iop    ___   n
   1____________________11_  11 ___  3       intl   ___   n
```

The table in our example shows 6 translations for calls that begin with 0.

Instructions

Let's use the ARS digit analysis table shown above and follow the routing for an operator assisted a call to NJ.

— A user dials 9 0 908 956 1234.

— The switch drops the ARS FAC (9 in our example), looks at the ARS Digit Analysis Table for 0, and analyzes the number. The switch:

- determines that more than 1 digit was dialed
- rules out the plan for 00, 01, and 011
- determines that 11 digits were dialed

— The switch routes the call to route pattern 1 as an operator assisted call

Defining Inter-exchange carrier calls

Let's look at how the switch routes an ARS call to an inter-exchange (long-distance) carrier (IXC). IXC numbers directly access your long-distance carrier lines.

IXC numbers begin with 1010, followed by three digits, plus the number as it is normally dialed including 0, 00, or 1+ 10 digits. These numbers are set up on your default translations.

Remember, the user dials 9 to access ARS, then the rest of the number.

Instructions

To see how your switch handles a call to an IXC:

1. Type **display ars analysis 1** and press ENTER.

   The AAR and ARS Digit Analysis Table screen starting with 1 appears.

```
                         ARS DIGIT ANALYSIS TABLE
                                Location:  all           Percent Full:    6
        Dialed            Total      Route    Call  Node  ANI
        String           Min Max    Pattern  Type  Num   Req
 1____________________11_ 11 ___ 3        intl  ___   n
 1010xxx______________7 _ 7 ____ 5        op    ___   n
 1010xxx0_____________8 _ 8 ____ 5        op    ___   n
 1010xxx0_____________18_ 18 ___ 5        op    ___   n
 1010xxx00____________9 _ 9 ____ 5        op    ___   n
 1010xxx01____________17_ 25 ___ 3        iop   ___   n
 2____________________7 _ 7 ____ 2        hnpa  ___   n
 3____________________7 _ 7 ____ 2        hnpa  ___   n
 4____________________7 _ 7 ____ 2        hnpa  ___   n
```

This table shows 5 translations for IXC calls.

When you use x in the Dialed String field, the switch recognizes x as a wildcard. The x represents any digit, 0 - 9. If I dial 1010, the next 3 digits will always match the x wild cards in the dialed string.

Use the ARS digit analysis table shown above and follow the routing for an IXC call to AT&T. 1010288 is the carrier access code for AT&T.

— A user dials 9 1010288 plus a public network number.

— The switch drops the ARS FAC (9 in our example), looks at the ARS Digit Analysis Table for 1010, and analyzes the number.

— The switch matches 288 with xxx and sends the call over route pattern 5.

Restricted area codes and prefixes

Certain area code numbers are set aside in the North American Numbering Plan. These numbers are 200, 300, 400, 500, 600, 700, 800, 877, 888, 900. You need to specifically deny calls made to area codes 200 through 900 (except 800 and 888).

You can also deny access to the 976 prefix, which is set aside in each area code for pay-per call services, if you do not want to incur charges. You can block 976 or any other prefix in all NPAs with a single entry in the digit analysis table. See ‘‘Using wild cards’’ on page 251 for more information.

Instructions

Set the 200 area code apart from other area codes 201 through 209. We use the digit analysis table 120 because it defines long distance calls that begin with 1 and all area codes from 200 through 209.

To deny long distance calls to the 200 area code:

1. Type **change ars analysis 120** and press ENTER.

   The AAR and ARS Digit Analysis Table screen beginning with 120 appears.

```
                              ARS DIGIT ANALYSIS TABLE
                                     Location:  all          Percent Full:    6
          Dialed          Total      Route    Call  Node  ANI
          String         Min Max    Pattern  Type  Num   Req
   120__________________11  11      4       fnpa  ___   n
   1200_________________11_ 11____  deny     fnpa  ___   n
```

The table in our example shows 2 translations for calls that begin with 120.

First, follow the routing for a long-distance call that begins with 120 and is allowed. The 120 translation handles all dial strings 1-201 through 1-209, and there are many matches.

— A user dials 9 120 plus 8 digits (the first of the 8 digits is not 0).

— The switch drops the ARS FAC (9 in our example), looks at the ARS Digit Analysis Table for 120, and analyzes the number. The switch determines that the call is a long-distance call and sends the call over route pattern 4.

Now let's follow a call that begins with the restricted area code 200. Only one string matches this translation.

— A user dials 9 1200 plus 7 digits.

— The switch drops the ARS FAC (9), and looks at the ARS Digit Analysis Table for 1200. The switch determines that the call type is deny and the call does not go through.

Using wild cards

You can use wild cards to help separate out calls to certain numbers. Remember, when you use the wild card x in the Dialed String field, the switch recognizes x as any digit, 0 - 9. For example, you can restrict users from making calls to a 555 information operator where you might incur charges.

Instructions

To prevent callers from placing calls to long-distance 555 information numbers:

1. Type **change ars analysis 1** and press RETURN.

   The AAR and ARS Digit Analysis Table screen beginning with 1 appears.

```
                         ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6
       Dialed         Total      Route    Call  Node  ANI
       String        Min Max    Pattern Type  Num   Req
1___________________11_ 11 ___ 1        intl  ___   n
1xxx555_____________11_ 11 ___ 1        intl  ___   n
1010xxx_____________7 _ 7 ____ 1        op    ___   n
```

2. Use the arrow keys to move to a blank Dialed String field.
3. Enter **1xxx555** in the Dialed String field.
4. Enter **11** in the Total Mn and **11** in Total Mx fields.
5. Enter **deny** (denied) in the Route Pattern field.
6. Enter **fnhp** in the Call Type field.
7. Press ENTER to save your changes.

Defining local information calls

You can set up your switch to allow calls to local information, or 411.

Instructions

To allow 411 service calls:

1. Type **change ars analysis 4** and press RETURN.

   The AAR and ARS Digit Analysis Table screen beginning with 4 appears.

```
                         ARS DIGIT ANALYSIS TABLE
                                Location:  all          Percent Full:    6
       Dialed         Total      Route    Call  Node  ANI
       String        Min Max    Pattern Type  Num   Req
411_________________3_  3  ___ 1        svcl  ___   n
5___________________7 _ 7 ____ 2        hnpa  ___   n
6___________________7 _ 7 ____ 2        hnpa  ___   n
7___________________7 _ 7 ____ 2        hnpa  ___   n
911_________________3   3 ____ 1        svcl  ___   n
```

2. Use the arrow keys to move to a blank Dialed String field.
3. Enter **411** in the Dialed String field.
4. Enter **3** in the Total Mn and **3** in Total Mx fields.
5. Enter **1** in the Route Pattern field.
6. Enter **svcl** (service call) in the Call Type field.
7. Press ENTER to save your changes.

Setting up multiple locations

When you define location numbers for the cabinets in a switch, you can create numbering plans and time zone and daylight savings plans that are specific for each cabinet. Choose the location you want to use as your main location, and offset the system clock time on each cabinet according to their its location. For more information about defining location numbers with the change cabinet command, refer to your system's Maintenance manual.

Before you start

Be sure the Multiple Locations field on the System-Parameters Customer-Options screen is set to **y**. If this field is set to **n**, contact your Avaya representative for more information.

Be sure your daylight savings rules are administered. Daylight Savings Rule numbers are located on the Daylight Savings Rule Screen

Each cabinet in a switch and each port network in the cabinet must be assigned a location number. Refer to the maintenance information for your switch for more information.

Instructions

For example, let's set up multiple locations for a switch with cabinets in Chicago and New York. Location 1 is assigned to the cabinet in Chicago, our main office, so Central Standard Time is used for our main location. Location 2 is assigned to the cabinet in New York. We'll define the numbering plan area (NPA) for the Chicago and New York locations, and set the time zone offset for NY to show the difference in time between Eastern Standard Time and Central Standard Time.

Tip:
*Type **list cabinets** to see the cabinet screen and a list of cabinets and their locations.*

To define locations for cabinets in Chicago and New York:

1. Type **change multiple locations** and press ENTER.

   The Locations screen appears.

```
                                  LOCATIONS

                ARS Prefix 1 required for 10-digit NANP calls?

Number     Name                TimeZone     Daylight-Savings          Number Plan
                               Offset       Rule                      Area Code

   1       Chicago________     + 00:00      1_                        312
   2       New York_______     - 01:00      1_                        212
   3       _______________     _ __:__      __                        ___
   4       _______________     _ __:__      __                        ___
   5       _______________     _ __:__      __                        ___
   6       _______________     _ __:__      __                        ___
   7       _______________     _ __:__      __                        ___
   8       _______________     _ __:__      __                        ___
   9       _______________     _ __:__      __                        ___
  10       _______________     _ __:__      __                        ___
  11       _______________     _ __:__      __                        ___
  12       _______________     _ __:__      __                        ___
  13       _______________     _ __:__      __                        ___
  14       _______________     _ __:__      __                        ___
```

2. Type **y** in the ARS Prefix 1 required for 10-digit NANP calls field.

   Our dial plan requires users to dial a 1 before all 10-digit (long distance) NANP calls.

3. Type **Chicago** in the Name field in the Number 1 row.

   Use this field to identify the location.

4. Type **+00:00** in the TimeZone Offset field in the Number 1 row.

   In our example, the system time and the Chicago location time are the same.

5. Type **1** in the Daylight Savings Rule field in the Number 1 row.

   In our example, daylight savings rule 1 applies to U.S. daylight savings time.

   Tip:
   *Use **display daylight-savings-rules** to see what rules have been administered on your switch.*

6. Type **312** in the Number Plan Area Code field in the Number 1 row.

   In our example, 312 is the local area code for Chicago, location 1.

7. Type **New York** in the Name field in the Number 2 row.

8. Type **-01:00** in the TimeZone Offset field in the Number 2 row.

   In our example, subtract one hour from the system clock in Chicago to provide the correct time for the location in New York.

9. Type **1** in the Daylight Savings Rule field in the Number 2 row.

   In our example, daylight savings rule 1 applies to U.S. daylight savings time, and both locations use the same rule.

10. Type **212** in the NANP field in the Number 2 row.

    In our example, 212 is the local area code for New York, location 2.

11. Press ENTER to save your changes.

Related topics

Refer to ‘‘Setting the system date and time’’ on page 39 for more information about how to set your system clock and specify the daylight savings rule for the location.

Refer to ‘‘Establishing daylight savings rules’’ on page 37 for more information about how to specify the dates and times of daylight savings rules.

Routing with multiple locations

When you set up multiple locations, you can define call routing that covers all locations as well as call routing specific to each individual location. Use your routing tables to define local routing for 911, service operators, local operator access, and all local calls for each location. Leave long-distance and international numbers that apply across all locations on the routing tables with location field set to **all**.

Before you start

Be sure the Multiple Locations field on the System-Parameters Customer-Options screen is set to **y**. If this field is set to **n**, contact your Avaya representative for more information.

AAR or ARS must be administered.

- For AAR, verify that either the Private Networking field or the Uniform Dialing Plan field is **y** on the System-Parameters Customer-Options screen.
- For ARS, verify that the ARS field is **y** on the System-Parameters Customer-Options screen.

Each cabinet in a switch must be assigned a location number. Refer to the maintenance information for your switch for more information.

Instructions

For example, let's use ARS to set up local call routing for two switch locations. Our Chicago switch is assigned to location 1, and our New York switch is assigned to location 2.

Our example shows a simple local dialing plan. Each location already contains location-specific routing tables. We'll use route pattern 1 for local service calls and route pattern 2 for local HNPA calls in the Chicago location.

Tip:
Create location-specific routing by assigning different route patterns for each location.

To define local calls for switches in Chicago and New York:

1. Type **change ars analysis location 1** and press ENTER.

   The AAR and ARS Digit Analysis Table screen for location 1 appears.

```
change ars analysis                                              Page  1 of  X

                                  ARS DIGIT ANALYSIS TABLE
                                        Location: 1___      Percent Full:  ___

     Dialed           Total     Route     Call    Node   ANI
     String          Min Max   Pattern    Type    Num    Reqd
2________________    7   7      2____     hnpa    ___     n
3________________    7   7      2____     hnpa    ___     n
4________________    7   7      2____     hnpa    ___     n
411______________    3   3      1____     svcl    ___     n
5________________    7   7      2____     hnpa    ___     n
555______________    7   7      deny_     hnpa    ___     n
6________________    7   7      2____     hnpa    ___     n
611______________    3   3      1____     svcl    ___     n
7________________    7   7      2____     hnpa    ___     n
8________________    7   7      2____     hnpa    ___     n
811______________    3   3      1____     svcl    ___     n
9________________    7   7      2____     hnpa    ___     n
```

2. Type the information for local dialed strings and service calls in each row on the screen.

   In our example, for location 1 (Chicago) local HNPA calls:

   - Type the appropriate digit in the Dialed String field
   - Type **7** in the Total Min field
   - Type **7** in the Total Max field
   - Type **2** in the Route Pattern field
   - Type **hnpa** in the Call Type field

In our example, for location 1 (Chicago) local service calls:

- Type the appropriate digits in the Dialed String field
- Type **3** in the Total Min field
- Type **3** in the Total Max field
- Type **1** in the Route Pattern field
- Type **svcl** in the Call Type field

3. Press ENTER to save your changes.
4. Type **change ars analysis 4 location 2** and press ENTER.

   The ARS Digit Analysis Table for location 2 appears.
5. Type in the local HNPA and service call routing information for New York.
6. Press ENTER to save your changes.

Related topics

Refer to ‘‘Automatic routing — general’’ on page 1383 for more information on ARS.

Refer to ‘‘AAR and ARS Digit Analysis Table’’ on page 559, ‘‘AAR and ARS Digit Conversion Table’’ on page 564, and ‘‘Toll Analysis’’ on page 1163 for general information on ARS administration. You can define location specific entries in addition to the global entries on these screens.

Modifying call routing

If your system uses ARS Digit Analysis to analyze dialed strings and select the best route for a call, you must change the digit analysis table to modify call routing. For example, you'll need to update this table to add new area codes or to restrict users from calling specific areas or countries.

Adding a new area code or prefix

A common task for system administrators is to configure their system to recognize new area codes or prefixes.

When you want to add a new area code or prefix, you look up the settings for the old area code or prefix and enter the same information for the new one.

Tip:
*Use **display toll xxx** (where xxx is the prefix you want to add) to see if the new area code or prefix number is set up as a toll call (y) or not. Some users may not be allowed to dial toll call numbers.*

Instructions

Let's add a new area code. When the California area code, 415, splits and portions change to 650, you'll need to add this new area code to your system.

Tip:
If you do not need to use 1 for area code calls, omit the ***1*** *in steps 1, 3, and 5 in our example. Also, enter* ***10*** *in the Total Min and Total Max fields (instead of 11) in step 6.*

To add this non-local area code:

1. Type **list ars route-chosen 14152223333** and press RETURN.

   You can use any 7-digit number after 1 and the old area code (415). We used 222-3333.

   The ARS Route Chosen Report screen appears.

```
                         ARS ROUTE CHOSEN REPORT

    Location: 1                      Partitioned Group Number:  1

     Dialed                Total       Route    Call      Node      Location
     String               Min  Max     Pattern  Type      Number
     141                  11   11      30       fnpa                all
```

2. Write down the Total Min, Total Max, Route Pattern, and Call Type values from this screen.

   In this example, the Total Min is `11`, Total Max is `11`, Route Pattern is `30`, and the Call Type is fnpa.

3. Type **change ars analysis 1650** and press RETURN.

   The AAR and ARS Digit Analysis Table screen appears.

```
                         ARS DIGIT ANALYSIS TABLE
                               Location:  all           Percent Full:    6
           Dialed          Total      Route    Call  Node  ANI
           String         Min Max     Pattern  Type  Num   Req
    1_____________________11   11     30       fnpa  ___   n
    167___________________11   11     30       fnpa  ___   n
    1800__________________11   11     30       fnpa  ___   n
    2____________________ 7    7      2        hnpa  ___   n
    3____________________ 7    7      2        hnpa  ___   n
    4____________________ 7    7      2        hnpa  ___   n
    5____________________ 7    7      2        hnpa  ___   n
    7____________________ 7    7      2        hnpa  ___   n
    8____________________ 7    7      2        hnpa  ___   n
    911__________________ 3    3      1        emer  ___   n
    976_________________ _11   11     deny     hnpa  ___   n
```

4. Move to a blank Dialed String field.

   If the dialed string is already defined in your system, the cursor appears in the appropriate Dialed String field, where you can make changes.

5. Enter **1650** in the Dialed String field.

6. Enter the minimum and maximum values from step 2 in the Total Mn and Total Mx fields.

   In our example, enter **11** in each field.

7. Enter the route pattern from step 2 in the Route Pattern field.

   In our example, enter **30**.

8. Enter **fnpa** in the Call Type field.

9. Enter the node number from step 2 in the Node Num field.

   For our example, leave the node number blank.

10. Press ENTER to save your changes.

To add a new prefix, follow the same directions, except use a shorter dial string (such as **list ars route-chosen 2223333**, where **222** is the old prefix) and a dial type of **hnpa**.

Tip:

If you change an existing area code for a network with multiple locations, be sure to change the Number Plan Area Code field on the Locations screen. Refer to *"Setting up multiple locations'' on page 253*.

Using ARS to restrict outgoing calls

ARS allows you to block outgoing calls to specific dialed strings. For example, you can restrict users from making international calls to countries where you do not do business, or in the U.S. you can restrict access to 900 and 976 pay-per-call numbers.

SECURITY ALERT:

To prevent toll fraud, deny calls to countries where you do not do business. The following countries are currently concerns for fraudulent calling.

country	code	country	code
Colombia	57	Pakistan	92
Ivory Coast	225	Peru	51
Mali	23	Senegal	221
Nigeria	234	Yemen	967

Instructions

To prevent callers from placing calls to Colombia (57):

1. Type **change ars analysis 01157** and press RETURN.

   You enter 011 (international access) and the country code (57). The ARS Digit Analysis Table screen appears.

2. Move to a blank Dialed String field.

   If the dialed string is already defined in your system, the cursor appears in the appropriate Dialed String field. Skip to step 5 to deny calls to this dialed string.

3. Enter **01157** in the Dialed String field.
4. Enter **10** in the Total Mn and **23** in Total Mx fields.
5. Enter **deny** (denied) in the Route Pattern field.
6. Enter **intl** in the Call Type field.
7. Press ENTER to save your changes.

Overriding call restrictions

You can use authorization codes to enable callers to override a station's calling privileges. For example, you can give a supervisor an authorization code so they can make calls from a phone that is usually restricted for these calls. Since each authorization code has its own COR, the system uses the COR assigned to the authorization code (and FRL assigned to the COR) to override the privileges associated with the employee's phone.

Note that authorization codes do not override dialed strings that are denied. For example, if your ARS tables restrict users from placing calls to Colombia, a caller cannot override the restriction with an authorization code.

Before you start

Verify that the Authorization Codes field on the System Parameters Customer Options screen is set to **y**.

SECURITY ALERT:

You should make authorization codes as long as possible to increase the level of security. You can set the length of authorization codes on the Feature-Related System Parameters screen.

Instructions

Let's create an authorization code 4395721with a COR of 2.

1. Type **change authorization-code 4395721** and press RETURN.

   The Authorization Code — COR Mapping screen appears.

2. In the AC field, type **4395721**.
3. In the COR field, enter **2**.
4. Press ENTER to save your changes.

```
                      Authorization Code - COR Mapping
              NOTE: 2 codes administered. Use 'list' to display all codes.

   AC        COR  AC        COR  AC        COR  AC        COR  AC        COR  AC       COR
   9260839   3   ________ __  ________ __  ________ __  ________ __  ________ __
   2754609   4   ________ __  ________ __  ________ __  ________ __  ________ __
   ________ __  ________ __  ________ __  ________ __  ________ __  ________ __
   ________ __  ________ __  ________ __  ________ __  ________ __  ________ __
   ________ __  ________ __  ________ __  ________ __  ________ __  ________ __
```

Defining ARS Partitions

Most companies want all their users to be able to make the same calls and follow the same route patterns. However, you may find it helpful to provide special calling permissions or restrictions to a group of users or to particular phones.

ARS partitioning allows you to provide different call routing for a group of users or for specific phones.

NOTE:
If you used partitioning on a prior release of Avaya MultiVantage and you want to continue to use partitioning, please read this section carefully. In this release of Avaya MultiVantage, partition groups are defined on the Partition Route Table. If you want to define routing based on partition groups, use the Partition Route Table. Partition groups are no longer defined on the Digit Analysis Table.

Before you start

Verify that Partitioning on the System Parameters Customer Options screen is y.

Verify that Time of Day Routing on the System Parameters Customer Options screen is n.

Setting up partition groups

Let's say you allow your employees to make local, long distance, and emergency calls. However, you have a lobby phone for visitors and you want to allow users to make only local, toll-free, and emergency calls from this phone.

To restrict the lobby phone, you modify the routing for a partition group to enable only specific calls, such as U.S. based toll-free 1-800 calls, and then assign this partition group to the lobby phone.

Instructions

To enable 1-800 calls for partition group 2:

1. Type **list ars route-chosen 18002221000** and press RETURN.

   You can use any 7-digit number following the 1800 to create an example of the dialed string.

   The ARS Route Chosen Report for partition group 1 appears.

```
                         ARS ROUTE CHOSEN REPORT

   Location :  1                    Partitioned Group Number: 1

       Dialed             Total     Route      Call      Node     Location
       String            Min  Max  Pattern     Type      Number

1800___________________   11 11     p1___       fnpa      _____      all
```

2. Record the route pattern for the selected dialed string.

   In our example, the route pattern for 1800 is p1. This indicates that the system uses the Partition Routing Table to determine which route pattern to use for each partition.

   NOTE:
   If there was a number (with no p) under Route Pattern on the Route Chosen Report, then all partitions use the same route pattern. You need to use the Partition Routing Table only if you want to use different route patterns for different partition groups.

3. Press CANCEL to return to the command prompt.

4. Type **change partition-route-table index 1** and press RETURN.

   The Partition Route Table screen appears. In our example, partition group 1 can make 1800 calls and these calls use route pattern 30.

```
                          Partition Routing Table

                              Routing Patterns
Route
Index        PGN 1   PGN 2   PGN 3   PGN 4   PGN 5   PGN 6   PGN 7   PGN 8
-----        -----   -----   -----   -----   -----   -----   -----   -----
  1          __30    ____    ____    ____    ____    ____    ____    ____
  2          ____    ____    ____    ____    ____    ____    ____    ____
  3          ____    ____    ____    ____    ____    ____    ____    ____
  4          ____    ____    ____    ____    ____    ____    ____    ____
  5          ____    ____    ____    ____    ____    ____    ____    ____
  6          ____    ____    ____    ____    ____    ____    ____    ____
  7          ____    ____    ____    ____    ____    ____    ____    ____
```

5. In the PGN2 column that corresponds to Route Index 1, type **30** and press ENTER.

   This tells the system to use route pattern 30 for partition group 2 and allow partition group 2 members to make calls to 1800 numbers.

Assigning a phone to a partition group

To assign an extension to a partition group, first assign the partition group to a Class of Restriction (COR), and then assign that COR to the extension.

Instructions

To assign a Class of Restriction (COR) to partition group 2:

1. Type **list cor** and press RETURN.

   The Class of Restriction Information screen appears.

```
            CLASS OF RESTRICTION INFORMATION

                  COR   COR Description
                  0
                    1    supervisor
                    2    telecommuting
                  3
```

2. Choose a COR that has not been used and press CANCEL.

   In our example, select **3**.

3. Type **change cor 3** and press RETURN.

   The Class of Restriction screen appears.

```
                          CLASS OF RESTRICTION
                  COR Number: 3
             COR Description: lobby___________________

                         FRL: 0                                 APLT? y
     Can Be Service Observed? n           Calling Party Restriction: none
   Can Be A Service Observer? n            Called Party Restriction: none
      Partition Group Number: _       Forced Entry of Account Codes? n
            Priority Queuing? n                 Direct Agent Calling? n
        Restriction Override: none        Facility Access Trunk Test? n
        Restricted Call List? n                 Can Change Coverage? n
      Unrestricted Call List? __ __ __ __ __ __ __ __ __ __
               Access to MCT? y            Fully Restricted Service? n
        Category For MFC ANI: 7            Hear VDN of Origin Annc.? n
            Send ANI for MFE? n_            Add/Remove Agent Skills? n
               MF ANI Prefix: ______       Automatic Charge Display? n
 Hear System Music on Hold? y   PASTE (Display PBX Data on Phone)? n
                       Can Be Picked Up By Directed Call Pickup? n
                                   Can Use Directed Call Pickup? n
```

4. Type a name for this COR in the COR Description field.

   In our example, type **lobby**.

5. Enter **2** in the Partition Group Number field.

6. Press ENTER to save your changes.

Now assign COR 3 to the lobby phone at extension 1234:

1. Type **change station 1234** and press RETURN.

   The Station screen for 1234 appears.

2. In the COR field, enter **3**.

3. Press ENTER to save your changes.

Setting up time of day routing

Time of Day Routing lets you redirect calls to coverage paths according to the time of day and day of the week. You need to define the coverage paths you want to use before you define the time of day coverage plan.

You can route calls based on the least expensive route according to the time of day and day of the week the call is made. You can also deny outgoing long-distance calls after business hours to help prevent toll fraud. Time of Day Routing applies to all AAR or ARS outgoing calls and trunks used for call forwarding to external numbers.

Before you start

AAR or ARS must be administered on your switch before you use Time of Day Routing.

- For AAR, verify that either the Private Networking field or the Uniform Dialing Plan field is **y** on the System-Parameters Customer-Options screen.
- For ARS, verify that the ARS field is y and the Time of Day Routing field is **y** on the System-Parameters Customer-Options screen.

Instructions

As an example, let's allow our executives to make long distance calls during business hours. Let's look at the Time of Day Routing Plan before we make any changes.

To display your Time of Day Routing Plan:

1. Type **display time-of-day 1** and press RETURN.

   The Time of Day Routing Plan 1 appears.

```
                                                         Page 1 of 1

                        TIME OF DAY ROUTING PLAN 1

      Act    PGN Act      PGN   Act      PGN   Act       PGN   Act      PGN Act
      Time    #   Time     #    Time      #    Time       #    Time      #  Time
 Sun 00:01   1 __:__      _    __:__     _    __:__      _    __:__     _ __:__  _
 Mon 00:01   1 08:__00    2    12:__00   1    13:__00    2    17:__00   1 __:__  _
 Tue 00:01   1 08:__00    2    12:__00   1    13:__00    2    17:__00   1 __:__  _
 Wed 00:01   1 08:__00    2    12:__00   1    13:__00    2    17:__00   1 __:__  _
 Thu 00:01   1 08:__00    2    12:__00   1    13:__00    2    17:__00   1 __:__  _
 Fri 00:01   1 08:__00    2    12:__00   1    13:__00    2    17:__00   1 __:__  _
 Sat 00:01   1 __:__      _    __:__     _    __:__      _    __:__     _ __:__  _
```

   Make a note of the routing plan that is currently in effect. In our example, this plan is for employees who can only make local calls.

   You can see that in our example, two partition group numbers control time of day routing. PGN 1 begins one minute after midnight (00:01) every day of the week, and is used for after-business hours and all day Saturday and Sunday. PGN 2 is assigned to office hours Monday through Friday, not including noon (12:00) to 1:00 p.m. (13:00).

2. Press CANCEL to clear the screen.

Now let's create a new time of day routing plan for long-distance calls for our executives.

1. Type **change time-of-day 2** and press RETURN.

   The Time of Day Routing Plan 2 appears.

2. Type **1** in each field as shown on Time of Day Routing Plan 1.

   In our example, this is the PGN used for after hours and the lunch hour.

3. Type **3** in all other fields.

   In our example, PGN 3 uses the route pattern for long-distance calls during business hours. We can save money by using the trunk lines provided by our new long-distance carrier.

4. Press ENTER to save your changes.

Now assign your new Time of Day Routing Plan 2 to the COR assigned to your executives. Refer to ''Class of Restriction'' on page 636 to see where to assign this field.

Example

For this example, assume the following:

- Jim is the user at extension 1234.
- Extension 1234 is assigned a COR of 2.
- COR 2 is assigned a Time of Day Plan Number of 1.
- The Time of Day Routing Plan 1 is administered as shown in the example above.

When Jim comes into work on Monday morning at 8:30 and makes an ARS call (dials the ARS access code followed by the number of the person he is calling), the system checks the Time of Day Plan Number assigned to Jim's COR.

Because Jim has a COR of 2 with Time of Day Plan Number 1, the system uses Time of Day Routing Plan 1 to route the call.

According to Time of Day Routing Plan 1, calls made between 8:00 a.m. and 11:59 a.m. route according to the route pattern set up on PGN 1.

If Jim makes a call between 12:00 p.m. and 1:00 p.m. on Monday, the Time of Day Routing Plan 1 is used again. However, this time the call is routed according to PGN 2.

Related topics

Refer to “Route Pattern” on page 1039 screens for more information.

Refer to “Defining ARS Partitions” on page 261 to see how to set up partition groups.

Managing multimedia calling

9

Multimedia Applications Server Interface

The Multimedia Applications Server Interface (MASI) defines a protocol and a set of operations that are used to extend Avaya MultiVantage feature functionality to a Multimedia Communications Exchange (MMCX) system. MASI architecture fits the client/server model, where the Avaya MultiVantage functions as a server for MMCX clients. Examples of features supported by MASI include Call Detail Recording (CDR), AUDIX/INTUITY voice mail integration, and AAR/ARS.

MMCX can make use of both MASI features and MMCX autonomous features. Autonomous features are those that MMCX provides, even if MASI is not enabled. This document does not discuss them unless there is a consideration for MASI administration.

Some autonomous MMCX features:

- Basic Call (Place/Drop)
- Call Coverage
- Conference
- Transfer

MultiVantage/MASI features:

- Basic Call (Place/Drop) — Avaya MultiVantage tracks the status of all calls placed to or from a MASI terminal.
- Call Detail Recording — Avaya MultiVantage tracks calls to and from MASI terminals and can produce call records that indicate if a call uses MASI.
- Call Coverage — Avaya MultiVantage tracks MMCX calls that are sent to coverage. A MultiVantage coverage path can contain both MASI terminals and MultiVantage stations.
- Conference — Avaya MultiVantage tracks conference calls that involve MASI terminals, if a MultiVantage station originates the conference. Conferences that involve MASI terminals and MultiVantage stations are voice-only. If the MultiVantage station originates the call, the caller can use the consultative form of conference or transfer.
- World Class Routing (AAR or ARS) — Calls from MASI terminals can take advantage of Avaya MultiVantage World Class Routing capabilities.
- Voice messaging access to AUDIX/INTUITY — MMCX users can take advantage of AUDIX voice messaging, and receive message waiting indication.
- MMCX trunking — By assigning MultiVantage trunk access codes to interfaces from the MMCX to other MMCXs or the PSTN, Avaya MultiVantage can monitor traffic over those interfaces.

Before you start

CAUTION:

Avaya MultiVantage offers a wide range of features, and MMCX users may want to take advantage of this. In some cases, these features will operate as expected. However, some features are not supported for use over the MASI link, and their behavior is unpredictable. You may cause harm to your system by attempting to use these features. The Interactions section contains a list of features, and lists those features that are absolutely not supported for use with MASI. If you administer features on the DO NOT ADMINISTER list, Avaya cannot be responsible for the result.

For purposes of administration, there are feature buttons and groups of users that you must not administer with MASI terminal extensions. There are also features that you simply cannot administer for a MASI terminal, because the software does not allow it.

About this document

The following section describes the Multimedia Applications Server Interface, and provides instructions on how to set it up, including administration and monitoring. It also includes a section on troubleshooting.

You need to use both the DEFINITY Server system administration terminal (SAT) and the MMCX administration terminal to administer MASI. This document describes what you need to do at the DEFINITY Server SAT. It also occasionally mentions administration that you must do at the MMCX administration terminal. For more detailed MMCX information, see the *MMCX Technical Reference*.

List of terms

This is a list of terms that are specific to MASI, or that have meanings in the context of MASI that are not standard.

- **chgmasi** — The command you use to administer MASI at the MMCX administration terminal.
- **Interserver** — Connections between MMCX terminals on different MMCX servers/nodes.
- **MASI domain** — A MASI domain consists of a MultiVantage system and one or more MASI nodes that share the same dial plan. That is, the extension numbers on the MMCX are known to MultiVantage, and fit in the MultiVantage dial plan.
- **MASI interworking** — MASI interworking refers to the completion of a voice connection within a MultiVantage system, involving at least one MASI terminal and a MASI path.
- **MASI link** — The connection between the MMCX and the Avaya MultiVantage.
- **MASI node** — A single MMCX server. You may connect more than one MASI node to a MultiVantage system. Each node has a separate number. This node number needs to be consistent whenever referring to a specific MMCX server.
- **MASI non-interworking** — MASI non-interworking refers to the completion of a call by MMCX, not involving a MASI path.
- **MASI path** — The ISDN B-channels between MMCX and Avaya MultiVantage in a MASI environment. Paths are used for voice and data connections between Avaya MultiVantage and MMCX.
- **MASI signaling link** — ISDN D-channel used to transport a new ISO protocol called the MASI protocol between the Avaya MultiVantage and the MMCX.

- **MASI terminal** — The MultiVantage system representation of MMCX terminals in a MASI environment.
- **MMCX interface** — PRI interface for connecting an MMCX server to other public, private or WAN switching systems or equipment that is part of the public network. Similar to a MultiVantage trunk group. These may include non-MASI trunks connecting the Avaya MultiVantage and the MMCX.
- **MMCX trunk** — The MultiVantage system representation of trunk or network facilities terminating on MMCX. For purposes of MASI, they are called “interfaces.”

Planning for MASI

Before you start to administer MASI, you should make a plan for how to do it. Among the configurations on the following pages, there is probably one that matches the configuration of your system fairly closely. You might want to either write on these pages, or draw up your own configuration. It may help you if you have already determined trunk group and signaling group numbers, unused extensions, and so on. The following are things you need to consider:

- Establish the dial plan on the MMCX to agree with that of the Avaya MultiVantage. If you use Universal Dial Plan and MMCX, you may need to make adjustments for the MMCX dial plan.
- Find unused extensions and trunk group numbers. You need:
  - one trunk group number for each ISDN-PRI connection to the MMCX
  - one signaling group number for each MASI node and an unused MultiVantage extension for the signaling group
  - one unused MultiVantage extension for the Near-End Path Termination number for all MASI Paths to this ECS. You can use the same number for all MASI nodes in the domain.
  - two unused MMCX extensions for the nearpath and tscnum arguments to the **chgmasi** command. This is the command you use to administer MASI on the MMCX.

MASI configurations

There are several ways to set up combinations of MASI nodes and DEFINITY servers.The following figures depict several possible configurations.

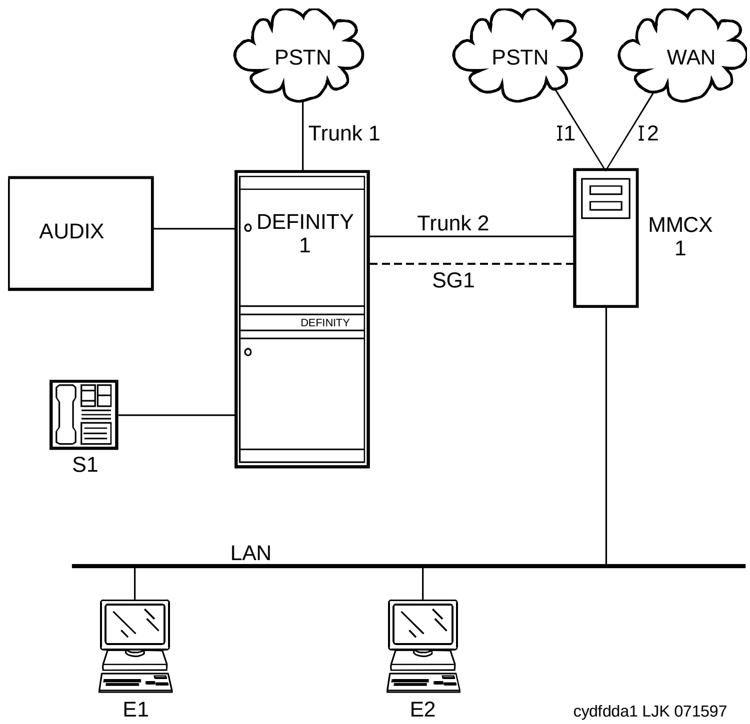


Figure 5. MASI domain of one Avaya MultiVantage running on a DEFINITY Server and one MMCX

The parts of this drawing, for MASI, are as follows:

- Trunk 1 — This is any type of trunk connection to the public network.
- Trunk 2 — This is the link between the Avaya MultiVantage solution and the MMCX, and requires a TN464C or later DS1 circuit pack. You administer this link as an ISDN-PRI trunk group, a MASI path and an NCA-TSC.
- I1 and I2 — These are MMCX interfaces to destinations other than Avaya MultiVantage solutions. Administer as MASI trunks.
- E1 and E2 — Endpoints (terminals) belonging to the MMCX. Administer as MASI terminals.
- MMCX — Determine a node number for each MMCX server. This can be any number from 1–15. Once established, Avaya MULTIVANTAGE informs the MMCX of its node number.
- S1 — MultiVantage station.

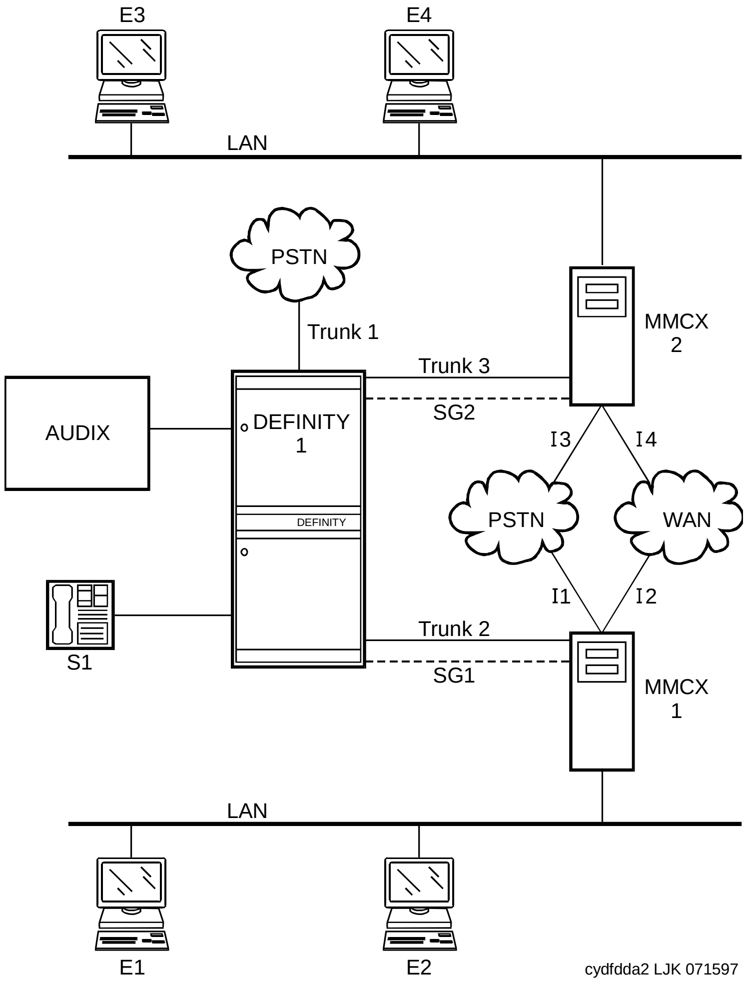


Figure 6. MASI domain of one Avaya MultiVantage running on a DEFINITY Server and two (or more) MMCXs

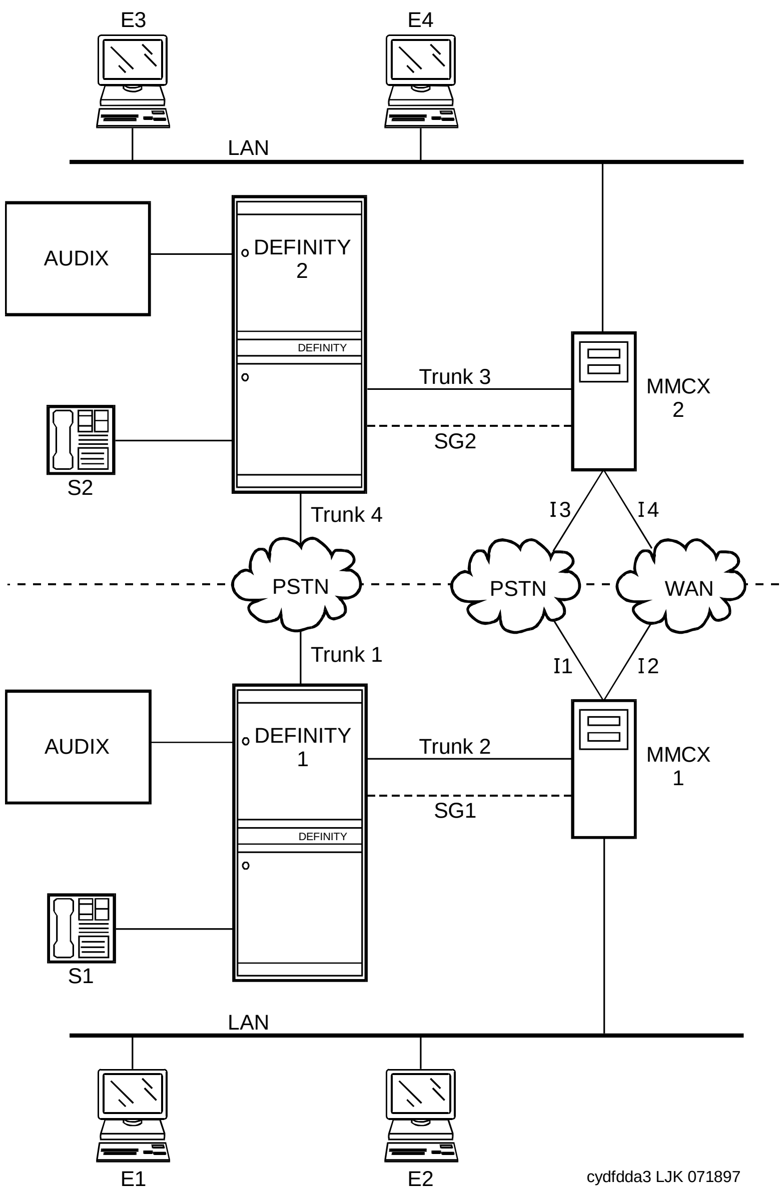


Figure 7. Two separate MASI domains

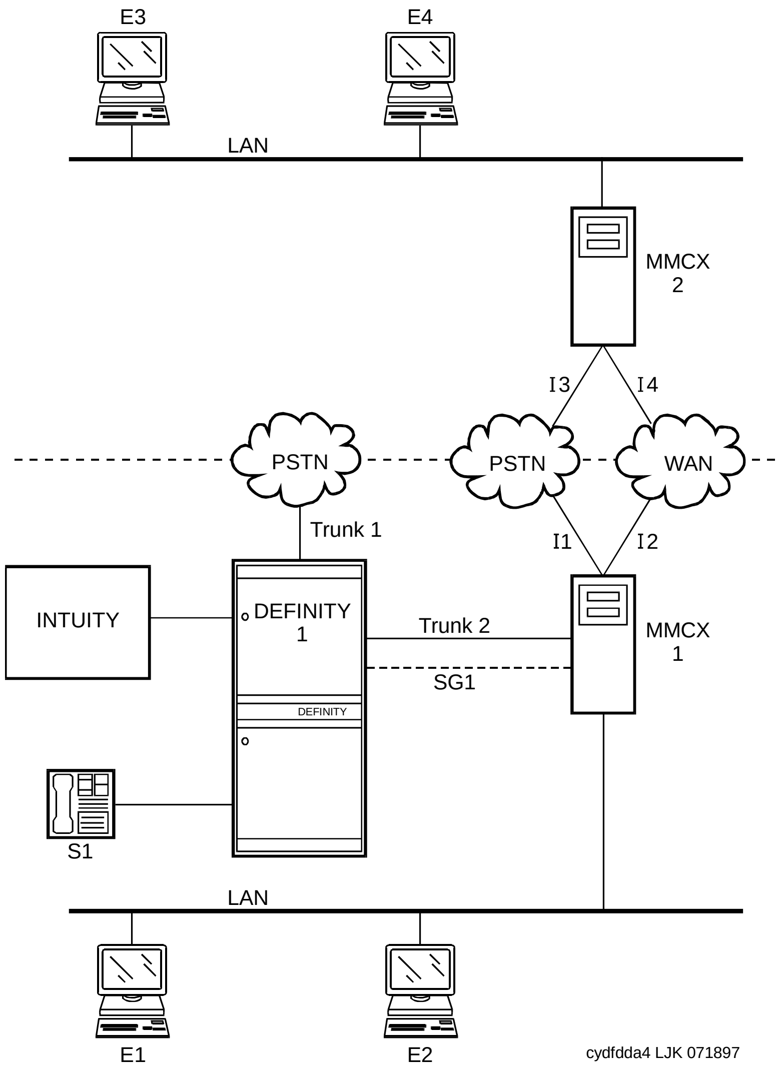


Figure 8. One MASI domain, and one non-MASI MMCX

The MASI node must be directly connected to the Avaya DEFINITY Server for MASI features to work. In this configuration, terminals that belong to MMCX 2 (E3 and E4), do not take advantage of MASI capabilities.

How to administer MASI

This section discusses the administration required to make MASI work. You perform most of this administration from the DEFINITY Server administration terminal. However, there are a few things you must do at the MMCX administration terminal. This section sometimes refers to the **chgmasi** command. This is the command you use to administer MASI parameters on the MMCX. For more information about using the chgmasi command, refer to your *MMCX Technical Reference*.

Step 1 — Establish customer options (Avaya)

An Avaya representative must activate MASI using the System-Parameters Customer-Options screen. The representative should also verify that ISDN-PRI (Avaya MultiVantage™ on a DEFINITY Server R configuration) or ISDN-PRI over PACCON (DEFINITY Server SI/DEFINITY Server CSI configurations), and AAR/ARS are enabled.

On the MMCX, MASI must be enabled using the **chgmasi** command.

Step 2 — Establish maintenance parameters and alarming options (Avaya)

Ensure that the Maintenance-Related System Parameters screen, Packet Bus Activated field = **y**.

Using the **set options** command (Avaya init or inads logins only), set MASI alarming options. For more information, see the *Avaya MultiVantage Maintenance and DEFNITY Server* for the appropriate model.

Step 3 — Establish the physical connection

Establish the physical connection between the Avaya DEFINITY Server and the MMCX. For more information about installing the cables, see *Installation for Adjuncts and Peripherals*.

Step 4 — Administer circuit pack

Using the DS1 Circuit Pack screen, verify that the DS1 circuit pack you use to establish the MASI link is administered as follows:

- Bit Rate = 1.544
- Line Coding = b8zs
- Line Compensation = 1
- Signaling Mode = isdn-pri

- Interface = network
- Country Protocol = 1
- Protocol Version = a

Step 5 — Administer a signaling group

For each MASI node, you need to establish a unique signaling group. Use the command **add signaling-group xxx** to access the Signaling Group screen.

```
                                                                 Page 1 of 5
                              SIGNALING GROUP
Group Number: _       Associated Signaling? y          Max number of NCA TSC: __
                         Primary D-Channel: ______      Max number of CA TSC: __
                                                      Trunk Group for NCA TSC: __
       Trunk Group for Channel Selection: ___
```

Screen 5. Signaling Group (Page 1 of 5)

For each link, establish a Non-Call Associated Temporary Signaling Connection (NCA-TSC) with the following attributes:

- **Associated Signaling** — MASI requires Facility Associated Signaling, so this field must be set to y.
- **Primary D-channel** — Enter a 6- to 7-character port number associated with the DS1 Interface circuit pack port. The port address of the PRI that carries D-channel signaling.

  The port number is used to assign the primary D-channel in the Signaling Group. For 24-channel facilities, the 24th port is assigned as the D-channel. For 32-channel facilities, the 16th port is assigned as the D-channel.
- **Max Number of NCA TSC** — For MASI, this must be 1.
- **Max number of CA TSC** — Leave the default of 0.
- **Trunk Group For NCA TSC** — This can be left blank.
- **Trunk Group for Channel Selection** — This can be left blank.

```
                                                              Page 2 of 5
                        ADMINISTERED NCA TSC ASSIGNMENT
Service/Feature: ___________           As-needed Inactivity Time-out (min):_
  TSC    Local                                                Adj.      Mach.
 Index    Ext.   Enabled  Established   Dest. Digits    Appl.   Name       ID
  1:     _____     __     _________    _____________   _______ _______     __
  2:     _____     __     _________    _____________   _______ _______     __
  3:     _____     __     _________    _____________   _______ _______     __
  4:     _____     __     _________    _____________   _______ _______     __
  5:     _____     __     _________    _____________   _______ _______     __
  6:     _____     __     _________    _____________   _______ _______     __
  7:     _____     __     _________    _____________   _______ _______     __
  8:     _____     __     _________    _____________   _______ _______     __
  9:     _____     __     _________    _____________   _______ _______     __
 10:     _____     __     _________    _____________   _______ _______     __
 11:     _____     __     _________    _____________   _______ _______     __
 12:     _____     __     _________    _____________   _______ _______     __
 13:     _____     __     _________    _____________   _______ _______     __
 14:     _____     __     _________    _____________   _______ _______     __
 15:     _____     __     _________    _____________   _______ _______     __
```

Screen 6. Administered NCA TSC Assignment page of the Signaling Group screen

- **Service/Feature** — Leave blank.
- **As-needed Inactivity Time-out (min)** — This field only applies to as-needed NCA-TSCs. Since MASI requires a permanent connection, leave blank.
- **TSC Index** — This display-only field specifies the administered NCA-TSCs assigned.
- **Local Ext** — Enter a valid, unassigned Avaya MultiVantage extension. This extension does not need a port assignment and does not need to correspond to any other administration.
- **Enabled** — Enter **y** to enable the administered NCA-TSC. You may want to wait to enable this link until all other administration is in place. If this is **y**, Avaya MultiVantage attempts to establish the connection as soon as you submit the form. This may cause your system to alarm, if other administration is not finished.
- **Establish** — Used to indicate the strategy for establishing this administered NCA-TSC. Enter **permanent** for MASI.
- **Dest. Digits** — A valid MMCX extension. This must correspond to the value of the *tscnum* argument to the **chgmasi** command.

NOTE:
These digits are sent as entered to the destination MMCX; no routing or other digit manipulation is performed.

- **Appl. —** Specifies the application this administered NCA-TSC is going to be used for. Enter **masi**.
- **Machine ID —** Used to indicate the MASI node to which this administered NCA-TSC is connected. This number should be the same as the MASI node number found on other screens.

How to list TSCs or determine status of the TSC

To determine which TSCs are designated for MASI, use the **list masi tsc** command.

```
                      MASI TEMPORARY SIGNALING CONNECTIONS (TSC)

Sig.  Primary   TSC      Local                                             Mach.
Grp   D-Chan    Index    Ext.   Enabled  Established   Dest. Digits      ID

xxx   xxxxxxx   xxx      xxxxx     x     xxxxxxxx      xxxxxxxxxxxxxxx   xx
xxx   xxxxxxx   xxx      xxxxx     x     xxxxxxxx      xxxxxxxxxxxxxxx   xx
xxx   xxxxxxx   xxx      xxxxx     x     xxxxxxxx      xxxxxxxxxxxxxxx   xx
```

Screen 7. MASI Temporary Signaling Connections (TSC) Display

This command displays the following:

- **Sig Grp** — The number of the signaling group to which this TSC belongs
- **Primary D-Channel** — Port location of the Primary D-channel
- **TSC Index** — The number of the MASI TSC within the signaling group
- **Local Ext.** — MultiVantage extension associated with the TSC
- **Enabled** — Indicates the state of the connection - enabled (y/n)
- **Established** — Value of established flag (as-needed/permanent)
- **Dest. Digits** — The MMCX extension that indicates the TSC destination
- **Mach. ID** — MASI node number

Once you establish and enable the signaling group, you need to verify that it is active. Use the command **status signaling-group signaling-group#** or **status tsc-administered signaling-group# [/tsc-index] [print]** to determine if the link is active.

Step 6 — Administer ISDN-PRI trunk group

Use the command **add trunk-group xxx** to access the Trunk Groups screen. For a more detailed description of the ISDN-PRI trunk group, see ''ISDN trunk group'' on page 894.

Establish an ISDN-PRI trunk group with the following attributes:

Page 1

- Group Type = isdn-pri
- TAC = valid TAC that conforms to your existing dial plan
- Direction = two-way
- Service Type = tie
- CDR Reports = n

You must also administer the PRI link from the MMCX to the ECS, using the MMCX administration terminal. See the *MMCX Technical Reference* for information on the **addpri** command.

Step 7 — Administer MASI Path Parameters

Use the **change masi path-parameters** command to access the MASI Path Parameters screen.

```
                              MASI PATH PARAMETERS

Near-End Path Extension: _____

    MASI Node       Trunk Group       Far-End Path Termination Number
        1               __                  __________________
        2               __                  __________________
        3               __                  __________________
        4               __                  __________________
        5               __                  __________________
        6               __                  __________________
        7               __                  __________________
        8               __                  __________________
        9               __                  __________________
       10               __                  __________________
       11               __                  __________________
       12               __                  __________________
       13               __                  __________________
       14               __                  __________________
       15               __                  __________________
```

Screen 8. MASI Path Parameters screen

Establish a MASI Path with the following attributes:

- **Near-End Path Extension** — An unassigned MultiVantage extension. When using the **chgmasi** command to administer the MMCX, this is the farpath extension. See the *MMCX Technical Reference* for more information.
- **MASI Node** — The node number for the MMCX. For each MMCX/MASI node, this number must be the same everywhere it occurs (signaling group, masi trunk group, and masi terminal screens).
- **Trunk Group** — This is the MultiVantage trunk group number for the ISDN-PRI trunk that will be used to establish call paths.
- **Far-End Path Termination Number** — This is an unassigned MMCX extension. When using the **chgmasi** command to administer the MMCX, this is the nearpath extension. See the *MMCX Technical Reference* for more information.

Step 8 — Administer MASI trunk groups

You use the MASI trunk group screen to define MMCX interfaces that interconnect MASI nodes, or that connect MMCX nodes to another private switch or central office. Examples of MMCX interfaces include:

- PRI trunks linking MMCX servers
- PRI trunks linking MMCX to the PSTN
- PRI trunks from MMCX to a MultiVantage system that are used for purposes other than MASI
- LAN interfaces linking MMCX servers

Use the command **add masi trunk-group xxx (or ‘next’)** to access the MASI Trunk Group screen. The trunk group number must not be assigned, and you cannot exceed the maximum total trunks for your system. Valid values for xxx are unused MultiVantage trunk group numbers between 1–96 for DEFINITY Server SI/DEFINITY Server CSI configurations, and 1–120 for DEFINITY Server R configurations.

```
                              MASI TRUNK GROUP

Group Number: 15                                                CDR Reports? y
  Group Name: INCOMING CALL______________ COR: 1_     TN: 1        TAC: 915_

                               MASI Node Number: __  Remote Group Number: _
```

Screen 9. MASI Trunk Group screen

- **Group Number** — This field displays the MASI trunk group number. This is the number assigned when executing the **add masi trunk-group** command.
- **CDR Reports** — Valid entries are **y**, **n**, and **r**. Default is **y**.
  - If you enter **y**, CDR records will be generated by completed outgoing calls terminated on this trunk group. If incoming calls are being recorded (the Record Outgoing Calls Only field on the CDR System Parameters screen is set to **n**), then a single CDR record will be generated for answered calls with the call duration.
  - If you enter **n**, no CDR records will be generated by calls originated by or terminated on this trunk group.
- **Group Name** — Enter a unique name that identifies the trunk group. Up to 27 characters can be used; default is “INCOMING CALL.”
- **COR** — Enter a class of restriction (COR) number (0–95) that reflects the desired restriction; default is “1.”
- **TN** — This field displays the Tenant Partition number. All MASI trunks are associated with Tenant 1.
- **TAC** — Enter the trunk access code (TAC) that identifies the trunk group on CDR reports. You must assign a different TAC to each MMCX interface. Valid entries conform to the dial plan (1–4 digits, * and # are valid first digits).
- **MASI Node Number** — The node number assigned to this MMCX machine.
- **Remote Group Number** — This is the number of the remote trunk group. For ISDN-PRI interfaces, valid values are any number 1–8; for LAN or WAN calling interfaces, the value must be 9. The combination of MASI Node Number and Remote Group Number must be unique. Remote group number corresponds to the group number on the MASI node.

How to view a list of all MASI trunk groups

To view a list of all the MASI trunks administered on the ECS, use the command **list masi trunk-group**.

```
                              MASI TRUNK GROUP

Group                                           Node    Remote
Number  TAC   Group Name                        Number  Grp No.  CDR  COR  TN

xxx     xxxx  xxxxxxxxxxxxxxxxxxxxxxxxxxx       xx      x        x    xx   xxx
```

Screen 10. List masi trunk-group output

How to determine the status of MASI trunk groups

To determine the status of a specific MASI trunk, use the command **status masi trunk-group xxx**, where xxx is the trunk group number. This command provides descriptive information about the trunk, and the number of currently active trunk calls.

```
                             MASI TRUNK GROUP STATUS

        Group Number: xxx      Number of Active MMCX Trunk Calls: xxx
    MASI Node Number: xx
 Remote Group Number: xxx
```

Screen 11. Status masi trunk-group output

Step 9 — Administer MASI terminals

Use the **add masi terminal xxxxx** or **next** command to administer each MASI terminal as a MASI terminal. You use available extensions on the ECS, so they need to conform to the Avaya MultiVantage dial plan. The extension must match the MultiVantage dial plan, and for the add command, the extension must not already be in use.The extension of the MASI terminal must match the number of the MASI terminal.

Avaya MultiVantage users dial the MASI Terminal Extension to reach MMCX users.

NOTE:
Anytime you add a terminal or other extension to the MMCX, you must administer a corresponding MASI terminal on the Avaya MultiVantage solution. If you do not, you will not be able to dial this extension from the Avaya MultiVantage solution.

```
                              MASI TERMINAL

Extension: 1000                                                 BCC: 0
                                          MASI Node Number: __ TN: 1__
                                                                COR: 1_
     Name: ___________________________


TERMINAL OPTIONS

       Send Display Info? y
```

Screen 12. MASI Terminal screen — page 1

- **Extension** — This field displays the extension that you entered on the command line.
- **BCC** — This field displays the bearer capability class of the terminal, and identifies the type of traffic the terminal supports. For MASI, this is always 0, for voice or voice-grade data.
- **MASI Node Number** — The number of the node on which this terminal resides.
- **TN** — The tenant partition in which this terminal resides. At present, all MASI terminals must reside within tenant 1. This field is display-only, and always 1.
- **COR** — The class of restriction associated with this terminal.

- **Name** — The name associated with the terminal. This can be any alphanumeric string up to 27 characters.
- **Send Display Info** — Indicates whether Avaya MultiVantage should forward display information associated with a call. Set to **y**.

```
                           MASI TERMINAL

FEATURE OPTIONS
           LWC Reception: none___

             CDR Privacy? n




              AUDIX Name: ________
```

Screen 13. MASI Terminal screen — page 2

- **LWC Reception** — This field indicates whether the terminal can receive Leave Word Calling messages. Valid values are none, audix, and spe (for DEFINITY Server R configurations) or mas-spe (for DEFINITY Server SI/DEFINITY Server CSI configurations). SPE-based LWC is not supported for MASI terminals. However, if embedded AUDIX is used without a Data Control Link, you must administer MASI terminals to receive SPE-based LWC messages. For such cases, the LWC feature is used by AUDIX to activate and deactivate message waiting lamps on MASI terminals.
- **CDR Privacy** — Indicates whether CDR Privacy is supported for this terminal. See ‘‘Call Detail Recording’’ on page 1437 for more information.

- **AUDIX Name** — This field only appears on DEFINITY Server R configurations. This field contains the name of the AUDIX adjunct for LWC messages. If LWC reception field is set to audix, this field must contain a name. The name must match a machine name on the Adjunct Names screen.

```
                              MASI TERMINAL
SITE DATA
       Room: _______
       Jack: _____
      Cable: _____
      Floor: _______
   Building: _______






BUTTON ASSIGNMENTS
 1: call-appr
```

Screen 14. MASI Terminal screen — page 3

- **Room** — Enter up to 10 characters to identify the MASI terminal location. This field may be blank.
- **Jack** — Enter up to 5 characters to identify the location where the MASI terminal is connected. This field may be left blank.
- **Cable** — Enter up to 5 characters to identify the cable that connects the MASI terminal to the system. This field may be left blank.
- **Floor** — Enter up to 7 characters to identify the floor where the MASI terminal is located.
- **Building** — Enter up to 7 characters to identify the building where the MASI terminal is located. Valid entries are listed in the site table.
- **Button Assignments** — This field contains a call appearance button and is display only.

Duplicate MASI terminal

Once you have one MASI terminal administered to your liking, you can use the **duplicate masi terminal** command to administer other stations with the same characteristics.

```
                             MASI TERMINAL


   Ext          Name                             Room      Jack  Cable
   77777        ___________________________      _________ _____ _____
   77778        ___________________________      _________ _____ _____
   77779        ___________________________      _________ _____ _____
   77781        ___________________________      _________ _____ _____
   77782        ___________________________      _________ _____ _____
   77783        ___________________________      _________ _____ _____
   _____        ___________________________      _________ _____ _____
   _____        ___________________________      _________ _____ _____
   _____        ___________________________      _________ _____ _____
   _____        ___________________________      _________ _____ _____
   _____        ___________________________      _________ _____ _____
   _____        ___________________________      _________ _____ _____
   _____        ___________________________      _________ _____ _____
   _____        ___________________________      _________ _____ _____
   _____        ___________________________      _________ _____ _____
```

Screen 15. Duplicate MASI Terminal screen

How to list and status MASI terminals

To view a list of all the MASI terminals administered on the ECS, use the command **list masi terminals**.This command only lists terminals within the domain of the Avaya DEFINITY Server from whose SAT you issue the command.

```
                             MASI TERMINALS


                                     Node
        Ext   Name                   Number CDR COR TN

        xxxxx xxxxxxxxxxxxxxxxxxxxxxxxxxx xx    x   xx  xxx
        xxxxx xxxxxxxxxxxxxxxxxxxxxxxxxxx xx    x   xx  xxx
        xxxxx xxxxxxxxxxxxxxxxxxxxxxxxxxx xx    x   xx  xxx
        xxxxx xxxxxxxxxxxxxxxxxxxxxxxxxxx xx    x   xx  xxx
```

Screen 16. List MASI Terminal output

To view the active or idle status of a specific MASI terminal, use the command **status masi terminal (extension)**.

```
                              GENERAL STATUS

            TYPE: MASI        Service State: active
       Extension: 54001
MASI Node Number: 14
```

Screen 17. Status MASI terminal command

To determine which extension you assigned as the MASI Near-End Path Termination extension, use the command **list extension-type**. This command displays the extension number and type (attendant, masi-terminal, etc.), as well as other information about the extension.

```
                              EXTENSION TYPE

                                                                     Cv1/
Ext    Type                 Name                     COR  TN  COS   Cv2
---    ----                 ----                     ---  --  ---   ----
1234   masi-terminal                                 1    1   1
4077   term-masi-path-call
```

Screen 18. List extension type

Step 10 — Administer features

AAR/ARS

1. AAR/ARS is an optional feature on Avaya MultiVantage, and you need to purchase this option to use it with MMCX. To verify that this feature is enabled, use the command **display system-parameters customer-options**. If it is not enabled, contact your Avaya representative.
2. The MMCX dial plan must use the same feature access codes as Avaya MultiVantage. If this is not already the case, modify the MMCX dial plan using the **chgdp** command. See the *MMCX Technical Reference* for more information.
3. Include this feature access code in the **chgmasi** command.

CDR

1. To get call detail records for calls over MMCX interfaces, set CDR Reports = **y** on the MASI Trunk Group screen.
2. To get call records for calls over the ISDN-PRI trunk group, set CDR Reports = **y** on the ISDN-PRI Trunk Group screen.
3. To track calls between a MASI terminal and other MASI terminals or MultiVantage stations, enter the MASI terminal extension on the Intraswitch CDR screen.
4. Enter n in the Record Non-Call Assoc TSC field on the CDR System Parameters screen.

NOTE:
If you use the same PRI trunks for MASI and non-MASI calls, it is strongly recommended that you do not enable CDR for these calls. Establish a separate trunk group for non-MASI calls and set CDR Reports = **n**.

Coverage

To establish coverage from a MASI terminal to AUDIX:

1. Use the MMCX user interface to enter the AUDIX hunt group extension as the coverage point. You cannot use MultiVantage coverage administration for MASI terminals.
2. If AUDIX ports are not administered on MultiVantage, do so.
3. Set up the MASI terminal as an AUDIX subscriber. Enter the MASI terminal extension in the extension field on the Subscriber Administration screen.

To establish coverage from a MASI terminal to another MMCX terminal or MultiVantage station:

1. Use the MMCX user interface to enter the desired extension as the coverage point for the MASI terminal. You cannot use MultiVantage coverage administration for MASI terminals.

Step 11 — Verify administration

Make test calls from MultiVantage to MMCX, ensure that you can indeed place and receive calls.

Call an unattended MASI terminal. Verify that the call goes to AUDX. Retrieve the call from the MASI terminal. Verify that all works as expected.

MASI command permissions

If you are the super-user for your system, you can restrict other administrative logins from changing MASI administration. To do this, use the **change permissions (login-ID)** command. Enter **y** in the Additional Restrictions field, then move to the Restricted Object List page of the screen.

You may restrict the following MASI-related objects:

- masi-path-parameters
- masi-terminal
- masi-trunk-group
- masi-tsc

Detailed description of features

AAR/ARS

MMCX can take advantage of advanced routing features for voice-only calls to the PSTN or a DEFINTY system private network. Users must enter the AAR/ARS access code before the rest of the dialed digits. MASI will route the call over the MultiVantage private network (AAR) or the public network (ARS), based on the digits supplied by the MMCX user.

Routing patterns must contain only trunk groups that actually terminate on the Avaya MultiVantage solution. Calls from one MMCX to another do not use AAR/ARS. Authorization codes are not supported.

Call Detail Recording

Using the MASI link, MultiVantage is able to track call detail information for calls made using MMCX terminals and interfaces. CDR records all calls originating from or terminating at a MASI terminal. MASI CDR does not record ineffective call attempts when all MASI paths are busy.

The Resource Flag value of 8 indicates a MASI call. This field appears in unformatted, int-isdn, expanded and customized CDR formats. For formats other than these, you can determine that a call involves a MASI terminal or trunk by the TAC, dialed number or calling number fields.

The following are the CDR capabilities of MASI. Administration information is under the heading ‘‘How to administer MASI’’ on page 277.

- Incoming/Outgoing Trunk Call Splitting

  Call splitting does not produce separate records for MMCX calls that are transferred or conferenced.

- Intraswitch CDR

  You can administer intraswitch CDR to monitor MASI terminals. To do this, simply add the MASI terminal extension on the Intraswitch CDR screen. MultiVantage then monitors calls from MASI terminals to other MASI terminals, and calls between MASI terminals and MultiVantage stations.

- CDR Privacy

  You can administer a MASI terminal for CDR Privacy.

- Account Code Dialing and Forced Entry of Account Codes

  This is not supported for MASI terminals. Therefore, make sure the COR you assign does not force entry of account codes.

- Trunk CDR

  You can get call detail records for all incoming and outgoing calls made over MMCX interfaces.

Call redirection / Voice-messaging access

MMCX users can enter a MultiVantage extension, including an AUDIX hunt group, Callmaster agent, attendant console or telephone as their coverage point.

If AUDIX is established as the MASI terminal’s coverage point, the MASI terminal receives message waiting indication, and dials the AUDIX hunt group extension to retrieve messages. Once connected to AUDIX, operation for the MMCX user is the same as for a MultiVantage station user, including use of # to identify the extension, if desired.

NOTE:
It is not possible to determine the call coverage status of a MASI terminal.

MultiVantage tracks calls to MASI terminals that follow the autonomous coverage path from the MASI terminal. MMCX calls redirected to MultiVantage stations contain display information.

MASI terminals that dial AUDIX directly, or that place calls to MASI terminals that cover to AUDIX, do not receive ringback if all AUDIX ports are busy. Instead, these callers see a message that the called party is busy, and the call drops.

Transfer

MASI terminals cannot transfer calls to MultiVantage stations, and cannot transfer a call to another MASI terminal if the call involves a MultiVantage station.

Conferencing

Conferences can involve both MASI terminals and MultiVantage stations, and either one may initiate the conference. MultiVantage stations participate in such conferences in voice-only mode. If an MMCX user initiates a conference that involves MultiVantage stations, the conference will drop when the initiator drops from the call. If a MultiVantage station initiates the conference, that station may drop without affecting the other conferees.

Status tracking - terminals and trunks

Avaya MultiVantage tracks the active/idle status of all MASI terminals, and monitors traffic over MMCX interfaces.

Trunk groups

For MASI purposes, there are two kinds of trunk groups: the ISDN-PRI trunk groups that serve as paths for establishing calls between MultiVantage stations or trunks and MASI terminals or interfaces, and the remote trunks that are interfaces from the MMCX to other entities. Each MASI remote trunk group appears to the switch as a single unit, with no concept of members within the group.

NOTE:
You cannot test, busy out, or release MASI remote trunk groups, since you cannot dial a MASI remote trunk TAC from the Avaya DEFINITY Server. The TAC merely identifies the trunk to the switch for status and CDR.

You cannot administer MASI trunks as part of MultiVantage route patterns.

Interactions & Unsupported Features

We can generalize feature interactions to some extent. For example, since there are no buttons available to a MASI terminal, any feature that requires a button is also not available. MASI cannot support features that require the user to dial a trunk access code for a MASI remote trunk, or a feature access code other than AAR/ARS. The MMCX dial plan may contain only those feature access codes that are supported.

CAUTION:
DO NOT ADMINISTER the following features! The following features are not supported for use over the MASI link, and Avaya cannot be responsible for the results if you attempt to administer them.

Unsupported Call Center features

- ASAI — You must not administer a MASI terminal in an ASAI domain. MASI terminals and MMCX trunks are not monitored by ASAI. It may be possible for a MASI terminal to place a call to a MultiVantage station that is part of an ASAI domain. ASAI will not be blocked from controlling this call, but there may be unpredictable results. The same is true for calls originating from an ASAI domain terminating at MASI terminals, and for ASAI-monitored hunt groups that contain MASI terminals.
- Automatic Call Distribution — You must not include a MASI terminal extension as part of an ACD hunt group. You must not mix MASI administration with anything related to ACD, including Outbound Call Management and PASTE.
- Call Vectoring — You must not include MASI terminal extensions in any step of a vector.

Unsupported Basic features

- Bridged Call Appearances — You must not administer a bridged appearance that involves a MASI terminal.
- Call Coverage — You must not administer a MASI terminal in a MultiVantage station’s coverage path.
- Call Forwarding — You must not forward a MultiVantage station to a MASI terminal.
- Call Pickup — You must not administer a MASI terminal as part of a pickup group.
- Intercom — You must not administer MASI terminals as members of any type of intercom group.
- Manual Message Waiting — You must not administer a manual message waiting button (man-msg-wt) with a MASI terminal as the referenced extension.
- Manual Signaling — You must not administer a manual signaling button (signal) with a MASI terminal as the referenced extension.
- Night Service — You must not administer a MASI terminal as a night service destination.
- Pull transfer — MASI terminals cannot perform a pull transfer operation. You must not administer this feature on an ECS where MASI is active. This applies only in Italy.
- Station Hunting — You must not administer a MASI terminal as part of a station hunting path.
- Terminating Extension Groups — You must not administer a MASI terminal as part of a TEG.

Other interactions

The following section describes feature behaviors that may not be as expected, but that are not likely to be destructive.

Attendant features

- Dial Access to the Attendant — MASI terminals will be able to dial the attendant access code, if it is administered in the MMCX dial plan.
- Attendant Direct Extension Selection — Attendants are able to access MASI terminals via DXS buttons and busy lamp indicates status of the MASI terminal.
- Emergency Access to the Attendant — MASI terminals have emergency access using the attendant access code, if it is administered in the MMCX dial plan. However, off-hook alerting is not administrable.
- Attendant Intrusion — Attendants are able to activate intrusion towards MASI terminals.
- Attendant Override — Attendants are not able to activate override towards MASI terminals.
- Attendant Recall — MASI terminals cannot activate attendant recall.
- Attendant Remote Trunk Group Select — Attendants cannot use this feature to select MASI remote trunks.
- Attendant Return Call — Operates normally if a MASI terminal is the called party.
- Attendant Serial Call — Serial calls are denied if the calling party is an MMCX interface.
- Attendant Straightforward Outward Completion — The attendant is able to complete calls to MultiVantage trunks for MASI terminals.
- Attendant Through Dialing —The attendant can use Through Dialing to pass dial tone to MASI terminals.
- Attendant Timers — Attendant timers work the same no matter what kind of terminal is involved.
- Attendant Trunk Group Busy/Warning Indicators — You cannot administer Busy/Warning indicators for MASI trunks because they are not standard MultiVantage trunks. However, you can administer these indicators for the trunk group administered for MASI paths.
- Attendant Trunk Identification — The attendant is not able to identify the trunk name via button pushes.

Basic features

- Abbreviated Dialing — A MultiVantage station can enter an MMCX extension in an AD list. However, MASI terminals cannot use AD.
- Administered Connections — MASI terminals must not be the originator nor the destination of an administered connection.
- Automatic Callback — Automatic callback does not work towards a MASI terminal.
- Automatic Circuit Assurance — You must not administer a MASI terminal as an ACA referral destination. You cannot administer ACA for MASI remote trunks.
- Busy Verification of Terminals and Trunks — You cannot use Busy Verification for MASI terminals or remote trunks.
- Call Detail Recording — CDR Account Code Dialing and Forced Entry of Account Codes are not supported for MASI terminals. See Call Detail Recording in Detailed Description for more information.
- Call Park — The attendant can park calls at the extension of a MASI terminal, but users can only retrieve these calls from a MultiVantage station, since MASI terminals cannot dial the Answer Back FAC.
- Data Call Setup — MultiVantage users cannot place data calls to MASI terminals.
- Facility Busy Indication — You can use FBI to track the status of MASI terminals. The FBI button and indicator lamp must be on a MultiVantage station. You cannot use FBI to track MMCX interfaces.
- Facility Test Calls — MultiVantage users cannot make test calls to MMCX interfaces.
- Go to Cover — MASI terminals cannot activate this feature.
- Leave Word Calling — The only valid LWC destination for a MASI terminal is AUDIX. You cannot administer SPE-based LWC. MASI terminals cannot send LWC messages to MultiVantage stations or MASI terminals.
- Loudspeaker paging — You can administer a MASI terminal as a code calling extension.
- Malicious Call Trace — MASI terminals cannot initiate malicious call trace.
- Message Retrieval — MMCX users can only retrieve messages through AUDIX.
- Music on Hold — Music on hold will only be available if a MultiVantage station has placed the call on hold.

- Override — Executive override does not work towards MASI terminals.
- Priority Calling — Priority calling is not supported for calls to or from MASI terminals.
- Ringback Queueing — Ringback Queueing is not supported for MASI terminals.
- Send All Calls — MMCX has an autonomous SAC function. See Call Redirection for more information.
- Tenant Partitioning — All MASI terminals exist in tenant 1, and you cannot change the tenant number.
- Time of Day coverage — As with all coverage, MultiVantage does not control coverage of the MASI terminal.
- Transfer out of AUDIX — A MASI terminal cannot use *T to transfer from AUDIX to another MASI terminal.

Hospitality features

- Do Not Disturb — MASI terminals cannot activate Do Not Disturb.

Multimedia features

- Multimedia Call Handling — Avaya MMCH users are not able to make H.320 calls to MASI terminals over the MASI link. Calls between MMCX terminals and MMCH terminals are voice only.

Troubleshooting

Verify proper operation using the following commands and follow normal escalation procedures to resolve any failures detected by the demand test.

1. Verify the DS1 trunk using the **test board <board location> long** command.
2. Verify the ISDN Signaling Group using the **test signaling-group <group number>** command. Also verify proper administration.
3. Verify the temporary signaling connection using the **test tsc-administered <group number>** command. Also verify proper administration.

Common error conditions

If the cable from the ECS to the MMCX becomes disconnected, you should see alarms raised against ISDN-SGRP and UDS1-BD. In particular, you should observe ISDN-SGRP errors such as 769, 1793, and 257. To resolve, reconnect the cable and follow normal test procedures.

If the far-end path termination number is incorrect, you should observe MASI-PTH error 513. To resolve, correct administration using the MASI Path Parameters screen.

If the Layer 3 TSC is not administered properly or is out of service, you should observe errors (but no alarms) raised against TSC-ADM. Verify the signaling group administration and follow normal escalation procedures for TSC-ADM. See the appropriate Avaya MultiVantage Maintenance manual for more information.

If the TSC fails to come up even through Layer 2 Signaling Group and below pass tests, you may run **test tsc-administered <group number>** to force a switch heartbeat test, or simply wait 5–10 minutes for the link to recover. This situation may happen if the switch is rebooted or if the MASI interface is administered before the MMCX is properly administered.

You may want to use busy port and release port commands to unlock things if features are not working.

Multimedia Call Handling

Multimedia Call Handling (MMCH) enables users to control voice, video, and data transmissions using a telephone and PC. Users can conduct video conferences and route calls like a standard voice call. They can also share PC applications to collaborate with others working from remote sites.

Operations in Basic or Enhanced modes

There are two distinct levels of functionality: Basic and Enhanced. The Basic mode of operation treats a standard-protocol H.320 multimedia call as a data call. If the call is redirected, it is converted to a voice call. As a voice call, certain features are enabled, such as coverage, voice mail, and multiparty video conferencing.

The Enhanced mode of operation allows a multifunction telephone to control a multimedia call as if it were a standard voice call. Spontaneous video conferencing, call forwarding, call coverage, hold, transfer and park, along with many routing features, are available to multimedia calls. Both modes of operation allow data collaboration between multiple parties using the T.120 standard protocol.

Definitions: MMCH features and components

Multimedia call

A multimedia call, for MMCH, is one that conforms to the H.320 and T.120 suite of protocol standards. These standards allow video-conferencing packages from different vendors to communicate with one another. The capabilities of the individual multimedia-endpoint package can vary, however.

- An H.320 call can contain voice, video and data.
- The bandwidth for MMCH calls is limited to 2 B-channels.

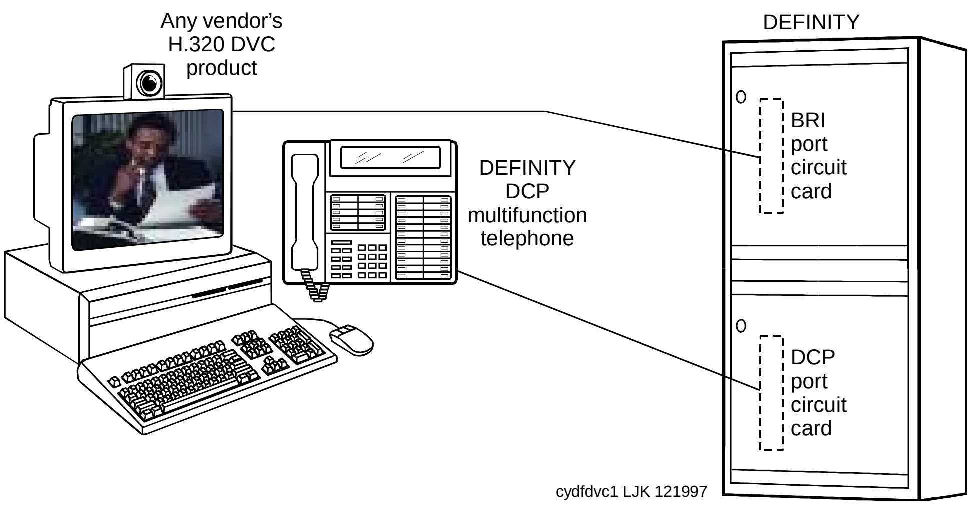


Figure 9. MMCH multimedia complex

Basic multimedia complex

A Basic multimedia complex consists of a BRI-connected multimedia-equipped PC and a non-BRI-connected multifunction telephone administered in Basic mode. With a Basic multimedia complex, users place voice calls at the multifunction telephone and multimedia calls from the multimedia equipped PC. Voice calls will be answered at the multifunction telephone and multimedia calls will alert first at the PC and, if unanswered, will next alert at the voice station. A Basic multimedia complex provides a loose integration of the voice station and H.320 DVC system.

Enhanced multimedia complex

An Enhanced multimedia complex consists of a BRI-connected multimedia-equipped PC and a non-BRI-connected multifunction telephone administered in Enhanced mode. The Enhanced multimedia complex acts as though the PC were directly connected to the multifunction telephone. Thus, voice call control, multimedia call control and call status are enabled at the telephone. An Enhanced multimedia complex provides a tight integration of the voice station and H.320 DVC system.

Multimedia endpoint

The multimedia endpoint is a user’s PC that has been equipped with an H.320 multimedia package. The PC is physically connected to the Avaya MultiVantage solution with a BRI line.

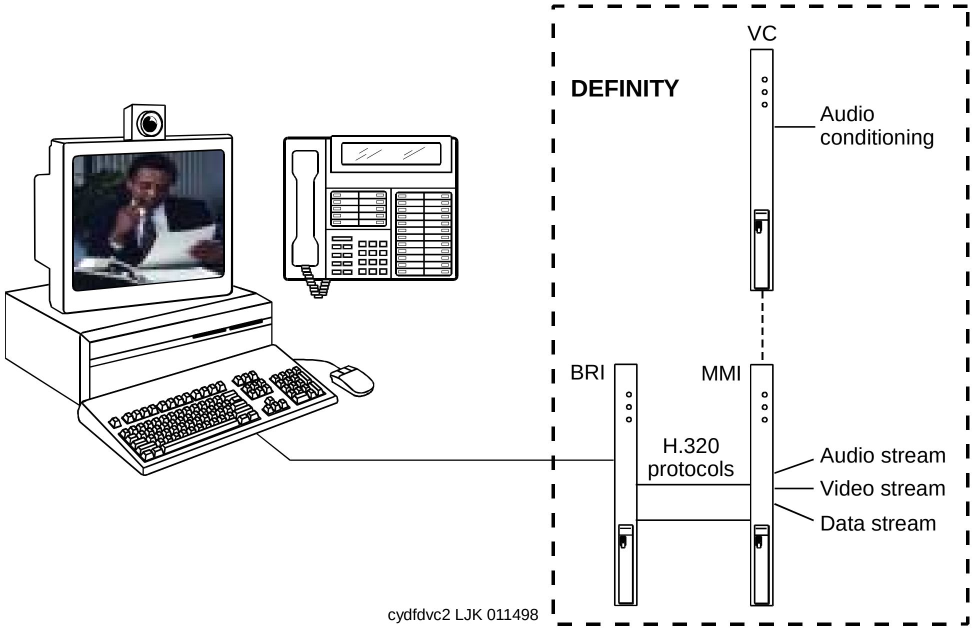


Figure 10. Enhanced MMCH service link

Enhanced mode service link

The service link is the combined hardware and software multimedia connection between the user's multimedia endpoint and the Avaya DEFINITY Server which terminates the H.320 protocol. The service link provides video, data, and, optionally, voice streams to augment the capabilities of the telephone and PC. A service link only applies to an Enhanced multimedia complex, never to a Basic multimedia complex. The service link is administered on the station screen and can be either "permanent" or "as-needed."

Feature Description

MMCH's two levels of functionality for a multimedia complex, Basic and Enhanced mode, are enabled either by switch administration or by an mm-basic feature button or FAC.

Basic Mode Operation

In Basic Mode:

- All voice-only calls originate at the voice station.
- All multimedia calls originate with the H.320 DVC system.
- All incoming voice calls attempt to alert at the voice station and receive all standard voice call treatment.
- All incoming H.320 multimedia calls attempt to alert on the H.320 DVC system initially. If answered, a 2-way video call will result. The Basic multimedia complex voice station will not be involved in the call in any way.

  If the H.320 multimedia call is not answered at the H.320 DVC system and the Basic multimedia complex voice station has the H.320 field administered to "y", the call will:

  — timeout at the DVC system

  — alert at the associated voice station set as a voice-only call

  — receive all standard voice call treatment.
- Call control depends on what type of call is being originated.

  — Video is received and controlled at the PC.

  — Voice is received and controlled at the telephone set.
- The voice station of a Basic multimedia complex must manually add their multimedia endpoint to a multimedia conference. There is limited support for multimedia feature interactions. A specific set of voice features work for multimedia calls.

- Service Links are not used by Basic mode complexes.
- A single number may be used to reach the Basic multimedia complex for voice or H.320 multimedia calls.

Enhanced Mode Operation

The Enhanced multimedia complex provides a much more tightly coupled integration of the complex voice station and H.320 DVC system. In Enhanced Mode:

- Both multimedia and voice calls must originate at the telephone set.
- Voice and multimedia calls can be controlled at the telephone set.
- Conferencing is spontaneous and established just like a voice-only conference call.
- There is extensive support for multimedia feature interaction. Most voice features work the same for multimedia calls.
- Service Links can be either “permanent” or “as-needed.”

Physical Installation

The physical components necessary to utilize MMCH capabilities include:

- H.320 DVC systems that are BRI connected to the Avaya DEFINITY Server.
- Non-BRI multifunction telephones.
- Avaya TN787 MultiMedia Interface (MMI) and TN788 Voice Conditioner (VC) boards.
- A T.120 Enhanced Services Module (ESM) server (necessary only if you plan to do T.210 data collaboration). Connectivity of the ESM requires an additional TN787 along with a TN2207 DS1 circuit pack.

Dual Port Desktop

Both Basic and Enhanced multimedia complexes are dual-port desktops that consist of:

- a BRI-connected multimedia-equipped PC that supports the H.320 protocol
- a non-BRI-connected multifunction telephone set.

The PC and the multifunction telephone are individually wired to the Avaya DEFINITY Server. These two pieces of equipment can be administratively associated to form a Basic or ENHANCED multimedia complex

MMCH works with any H.320 system that is fully H.320 compliant and operates at the 2B or 128K rate.

NOTE:
If you intend to share applications among users or whiteboard capabilities, the endpoint software you choose must also support the T.120 protocol.

The following endpoint-software packages have been tested:

- PictureTel PCS 50 & PCS 100, Release 1.6T
- Proshare 2.0a, 2.1
- Zydacron Z250 Ver. 2.02, Z350 Ver. 1.2 (With Netmeeting 2.0)

MMI & VC hardware

The MMCH feature requires the use of two additional circuit packs:

- Multi Media Interface (MMI) TN787J.
- Voice Conditioner (VC) TN788B.

The TN787 and TN788 are service circuit packs. The TN787 supports simultaneous operation of 16 2B H.320 calls. The TN788 supports the voice processing for 4 H.320 endpoints.

- These service circuit packs may be located in any Port Network.
- These packs do not require any translations as part of their implementation.
- The MMI and VC circuit packs are resource circuit packs akin to the Tone Detector circuit packs.
- These circuit packs require no switch administration and may be located in multiple port networks.
- Specific provisioning guidelines for the number and placement of these packs can be found by calling the Avaya Technical Support Center (TSC) at (303) 850 - 8187.

T.120 Data Collaboration Server

The Expansion Services Module (ESM) provides T.120 data collaboration capability on a MMCH multipoint H.320 video conference.

- Each person in the conference who wants to participate in the data collaboration session, must have a personal computer with an H.320 video application that supports the T.120 protocol.
- The Avaya DEFINITY Server must have an ESM installed.

ESM Installation

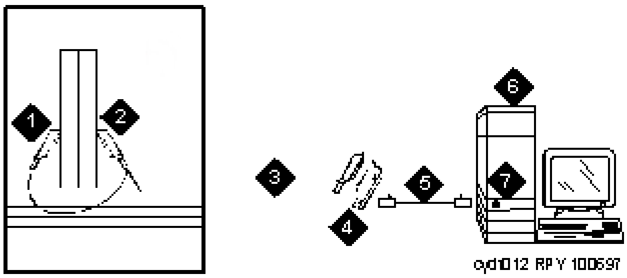


Figure Notes

1. Port B Y-cable connector to a TN787 multimedia interface (MMI) circuit pack
2. Port A Y-cable connector to a TN2207 PRI circuit pack
3. 25-pair Y-cable
4. 356A adapter
5. D8W cord connected to 356A adapter S/B port 8
6. Expansion service module (ESM)
7. Port B on compatible primary rate interface (PRI) card

Figure 11. Typical Multimedia Call handling ESM Connections

Use the following procedure and Figure 11 to connect to the ESM equipment:

1. Install the TN2207 primary rate interface (PRI) circuit pack and the TN787 multimedia interface (MMI) circuit pack in the MultiVantage System port carrier.

   NOTE:
   These two circuit packs should be co-located in the cabinet since they must be connected by a Y-cable on the back plane of the Avaya DEFINITY Server.

2. Record the circuit pack locations.
3. Connect the ESM Y-cable as shown.
4. Administer the DS1 screen and the signaling-group screen for the ESM (see ''ESM T.120 Server Administration'' on page 315).
5. Configure the ESM adjunct.

Planning for MMCH

The following are some of the tasks you perform in planning and administering MMCH.

Planning the system

This is a list of questions to help you use Avaya MultiVantage for multimedia.

- How many MMCH users are you going to have?
- How many multimedia calls do you expect to have at any given time?

With the information above you can determine how many Voice Conditioner (VC) and Multimedia Interface (MMI) circuit packs you need.

- Will users need data collaboration capabilities? If so, you need to install the Expansion Service Module (ESM).
- Which stations, hunt groups or vectors need early answer?
- Do you have ISDN-PRI trunks? It is possible to use separate DS1 trunks for data, but ISDN-PRI trunks are recommended.

Installation checklist

1. Purchase MMCH right-to-use.
2. Avaya — enable MMCH on System Parameters Customer-Options screen.
3. Administer default multimedia outgoing trunk parameter selection on the Feature-Related System-Parameters Features screen.
4. Administer MMCH related feature access codes on the Feature Access Code screen.
5. Install and administer hardware:
   - Install MMIs, VCs and ESM.
   - Administer ESM to ECS connection — DS1 Circuit Pack and Signaling Group screens.
   - Establish maintenance parameters — Maintenance-Related System Parameters screen.
6. Administer multimedia complexes:
   - Administer data modules — Data Module screen, or Data Module page of the Station screen.
   - Administer stations as part of a multimedia complex, assign associated data module extension, multimedia mode, service link mode and appropriate multimedia buttons — Station screen.

7. Administer early answer and H.320 flag for stations, the early answer flag for hunt groups, and the multimedia flag for vectors as appropriate.
8. Train end users.
9. Monitor traffic and performance.

Related screens

- System-Parameters Customer-Options
  - — Multimedia Call Handling (Basic)
  - — Multimedia Call Handling (Enhanced)
- Feature Related System-Parameters
  - — Default Multimedia Outgoing Trunk Parameter Selection (p.2)
- Maintenance-Related System Parameters
  - — Packet Bus Activated = y
  - — Minimum Maintenance Thresholds - MMIs, VCs
- Data Module (type = 7500 or WCBRI)
  - — Multimedia (p. 1) = y
  - — XID (p. 2) = n
  - — MIM Support (p. 2) = n
- Station
  - — MM Complex Data Ext (p. 1)
  - — H.320 Conversion (p. 2)
  - — Multimedia Early Answer (p. 2)
  - — Multimedia Mode (p.2)
  - — Service Link Mode (p.2)
  - — Feature Buttons (p.3) (optional)
- Hunt Group
  - — MM Early Answer (optional)
- Call Vector
  - — Multimedia (optional)

- Feature Access Codes
  - Basic Mode Activation (p.5)
  - Enhanced Mode Activation (p.5)
  - Multimedia Call Access Code (p.5)
  - Multimedia Data Conference Activation & Deactivation (p.5)

  The Multimedia Data Conference Deactivation FAC must be entered after you are active on a multimedia call. To enter the FAC:

  1. Select TRANSFER
  2. Receive a dialtone
  3. Dial the FAC
  4. Receive a confirmation tone
  5. Re-select the call appearance for the held multimedia call.

  - Multimedia Multi-Address Access Code (p.5)
  - Multimedia Parameter Access Code (p.5)
- DS1 Circuit Pack (ESM Only)
  - Bit Rate=2.048
  - Line Coding=hdb3
  - Signaling Mode=isdn-pri
  - Connect=pbx
  - Interface=network
  - Country Protocol=1
  - CRC=y
  - MMI Cabling Board
- Signaling group (ESM Only)
  - Primary D-Channel

Administration commands

System-Parameters Customer-Options screen:

To enable the MMCH release 6.3 feature, you must change the System-Parameters Customer-Options screen. There are two MMCH related fields on page 2 of this screen.

- The “Multimedia Call Handling (Basic)” field must be set to y to allow MMCH Basic mode operation.
- The “Multimedia Call Handling (Enhanced)” field must be set to y to allow MMCH Enhanced mode operations.

System-Parameters Features screen:

The default bandwidth for MMCH calls is defined on the System-Parameters Features screen shown below.

NOTE:
Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen.

- This default parameter will be either 2x56 or 2x64.
- The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

Data module screen

The H.320 DVC system should contain a BRI interface. You must connect this BRI interface to a port on a TN556 BRI circuit pack and administer it as a BRI data module.

- You may administer the data endpoint type as 7500 (recommended) or WCBRI.
- The fields for multimedia are the same on either screen.
- The administration for a Basic mode and an Enhanced mode data module are exactly the same.

```
                                                                    Page 1 of 2
                              DATA MODULE
   Data Extension: 65001     Name: ___________________________
             Type: 7500_______ COS: 1_                      Multimedia? y
             Port: _____       COR: 1_            MM Complex Voice Ext: 67434

ABBREVIATED DIALING
  List1: ____________
SPECIAL DIALING OPTION: hot-line
   HOT LINE DESTINATION
        Abbreviated Dialing Dial Code (From above list): _
CIRCUIT SWITCHED DATA ATTRIBUTES (used for modem pooling)
   Default Duplex: full     Default Mode: async     Default Speed: 1200_
DATA MODULE CAPABILITIES
  Default ITC: restricted                  Default Data Application? M2_A
```

Screen 19. Data Module screen (Page 1 of 2)

```
                                                                    Page 2 of 2
                              DATA MODULE
BRI LINK/MAINTENANCE PARAMETERS
                  XID? n      Fixed TEI? n      TEI: ___
         MIM Support? n
```

Screen 20. Data Module screen (Page 2 of 2)

- **Type** — Set the data module type to 7500 or WCBRI.
- **Multimedia** — This field appears on the data module screen only if MM is set to yes on the System-Parameters Customer Options screen. Enter **y** to enable this data module to be multimedia compliant.
- **MM Complex Voice Ext:** (display only) — This field contains the number of the associated telephone in the complex. This is a display-only field, and is blank until you enter the data module extension in the Station screen MM Complex Data Ext field. Once you have done that, these two extensions are associated as two parts of a multimedia complex.
- **XID and MIM Support** — Valid entries are **y** (default) and **n**. These fields must be set to n.

Station screen

After you have administered the BRI data module, use the Station screen to associate it with a voice station to screen a multimedia complex. This is a one-to-one relationship: you can administer only one station and one data endpoint per multimedia complex. Neither the voice station, nor the data endpoint can be a member of another multimedia complex.

NOTE:
A BRI station cannot be part of a multimedia complex.

- **H.320 Conversion** — **Valid entries are y and n (default).** This field is optional for non-multimedia complex voice stations and for Basic multimedia complex voice stations. It is mandatory for Enhanced multimedia complex voice stations. Because the system can only handle a limited number of conversion calls, you may need to limit the number of telephones with H.320 conversion. Enhanced multimedia complexes must have this flag set to **y**.

  For non-multimedia complex voice stations, setting this field to **y** allows H.320 calls to convert to voice and alert at the stand-alone voice station. If the call is unanswered at the voice station, the call will follow standard voice treatment. Any subsequent station that is reached in the routing of this call, that is, coverage points, forwarded destinations, call pickup members, and so forth, do not need to have the H.320 field enabled. The H.320 field is only needed at the ***first*** station that may receive the H.320 call.

  For Basic multimedia complex voice stations, setting this field to **y** allows H.320 calls to convert to voice and alert at the Basic multimedia complex voice station ***after*** an attempt has been made to offer the call to the H.320 DVC system. If the call is unanswered at the H.320 DVC system, the call will alert at the voice station after 5 seconds or after the administered number of rings as specified in the voice station's coverage path. If the call is unanswered at the voice station, the call will follow standard voice treatment. Any subsequent station that is reached in the routing of this call, that is, coverage points, forwarded destinations, call pickup members, and so forth, do not need to have the H.320 field enabled. The H.320 field is only needed at the ***first*** station that may receive the H.320 call.
- **Service Link Mode** — The service link is the combined hardware and software multimedia connection between an Enhanced mode complex's H.320 DVC system and the Avaya DEFINITY Server which terminates the H.320 protocol. A service link is never used by a Basic mode complex H.320 DVC system. Connecting a service link will take several seconds. When the service link is connected, it uses MMI, VC and system timeslot

resources. When the service link is disconnected it does not tie up any resources. The Service Link Mode can be administered as either ‘as-needed’ or ‘permanent’ as described below:

— As-Needed - Most non-call center multimedia users will be administered with this service link mode. The as-needed mode provides the Enhanced multimedia complex with a connected service link whenever a multimedia call is answered by the station and for a period of 10 seconds after the last multimedia call on the station has been disconnected. Having the service link stay connected for 10 seconds allows a user to disconnect a multimedia call and then make another multimedia call without having to wait for the service link to disconnect and re-establish.

— Permanent - Multimedia call center agents and other users who are constantly making or receiving multimedia calls may want to be administered with this service link mode. The permanent mode service link will be connected during the station’s first multimedia call and will remain in a connected state until the user disconnects from their PC’s multimedia application or the Avaya DEFINITY Server switch restarts. This provides a multimedia user with a much quicker video cut-through when answering a multimedia call from another permanent mode station or a multimedia call that has been early answered.

- **Multimedia Mode** — There are two multimedia modes, Basic and Enhanced, as described below:

— Basic — A Basic multimedia complex consists of a BRI-connected multimedia-equipped PC and a non-BRI-connected multifunction telephone set. When in Basic mode, users place voice calls at the multifunction telephone and multimedia calls from the multimedia equipped PC. Voice calls will be answered at the multifunction telephone and multimedia calls will alert first at the PC and if unanswered will next alert at the voice station if it is administered with H.320 = y. A Basic mode complex has limited multimedia feature capability as described in ‘‘Basic Mode Operation’’ on page 301).

— Enhanced — An Enhanced multimedia complex consists of a BRI-connected multimedia-equipped PC and a non-BRI-connected multifunction telephone. The Enhanced mode station acts as though the PC were directly connected to the multifunction telephone; the service link provides the actual connection between the Avaya DEFINITY Server and the PC. Thus, voice and multimedia calls are originated and received at the telephone set. Voice and multimedia call status are also displayed at the telephone set. An Enhanced mode station allows multimedia calls to take full advantage of most call control features as described in ‘‘Enhanced Mode Operation’’ on page 302.

- **Multimedia Early Answer** — Valid entries are **y** and **n** (default). This field lets you set this phone for early answer of multimedia calls. The system will answer the incoming multimedia call on behalf of the station and proceed to establish the H.320 protocol. After audio path has been established to the caller, the call will then alert at the voice station. The station may then answer by going off-hook and will have immediate audio path. No hourglass tone will be heard by the answering party (see ‘‘Hourglass Tone’’ on page 322).

  Example: An administrative assistant who does not have a multimedia PC, but may get multimedia mode calls from forwarding or coverage, may want to set the H.320 flag to **y** and the early answer flag to **y** on their voice station. This allows any multimedia call to be presented to the station with immediate voice path rather than hourglass tone. The answered call could then be transferred as voice to voice mail or transferred as multimedia to a user equipped with a multimedia endpoint.

Multimedia Buttons

There are six new multimedia specific buttons that may be added to a voice station. Most of them may be placed on any voice station, whether it is part of a Basic multimedia complex, an Enhanced multimedia complex or not part of any multimedia complex. Two feature buttons, mm-basic and mm-pcaudio, may only be placed on stations which are part of an Enhanced multimedia complex.

All of the multimedia specific feature buttons have a corresponding feature access code except mm-pcaudio and mm-cfwd.

- The mm-pcaudio feature can only be used via button.
- The mm-cfwd button may be replaced by the standard “*call forward*” FAC followed by the “*multimedia call*” FAC.

- **mm-call** — This button may exist on any voice station. Most multimedia enabled users will want an mm-call button. This button (or it's corresponding FAC) must be used to indicate that the user is placing a multimedia mode call. To place a multimedia mode call the user would go off-hook, select an idle call appearance, and press the mm-call button followed by the destination extension digits. If the user has a speakerphone they can simply press the mm-call button, which preselects an idle call appearance, followed by the destination extension digits.

  The mm-call button lamp lights when you press this button during call origination. The lamp also lights to indicate that the selected call appearance is a multimedia mode call.

- **mm-basic** — This button is only allowed on the voice station of a multimedia complex. The mm-basic button toggles a station between Basic and Enhanced modes. This button can NOT be used to change the station's multimedia mode when the station has an active multimedia call appearance.

  Toggling between Basic and Enhanced mode changes the station's administered Multimedia mode. When in Basic mode this field on the station screen will show `basic`. When in Enhanced mode this field on the station screen will show `enhanced`. The current station Multimedia mode will be saved to translation when a **save translation** command is executed.

- **mm-pcaudio** — This button only works for an Enhanced multimedia complex voice station. When originating or receiving a multimedia call, the audio path is connected to the voice station's handset or speakerphone device. The mm-pcaudio button allows a user to switch the audio portion of any call to their PC's audio input/output device (if available). To switch the audio path to the PC while active on a call, the user presses the mm-pcaudio button (if off-hook you may now hang up the handset). The mm-pcaudio button's status lamp will light up when the button is pushed to move the audio path to the PC and remain lit while the audio path is at the PC device.

  NOTE:
  If you are on a voice only call, the voice path will switch to the PC device but you will get muted or loopback video depending on the multimedia endpoint software.

  A user may simply go off-hook on their voice station or press the speakerphone button to move the audio path of a multimedia call from the PC back to the voice station. Pressing the mm-pcaudio button while the status lamp is lit and the voice station's handset is on-hook will disconnect the user from the active call.

- **mm-datacnf** — Pressing the mm-datacnf button from any voice station that is participating in a multimedia call will light the status lamp and alert the Avaya DEFINITY Server that you want to enable T.120 data collaboration with the other parties on the call. The button status lamp will also light for other participants in the multimedia call who have mm-datacnf buttons.

  Pressing this button from the voice station that enabled data collaboration on a multimedia mode call will deactivate the data session and revert to a voice and video call. If you are participating on a multimedia call with data collaboration, but did not initiate the data collaboration, and you press this button, the status lamp led will flash momentarily and the T.120 data services will not be terminated, (only the station that activated the collaboration session can deactivate it). This button only works for stations connected to an Avaya DEFINITY Server equipped with an ESM adjunct.

- **mm-cfwd** — The mm-cfwd button allows a user to indicate that multimedia mode calls will be forwarded as multimedia mode calls to a specific forwarded-to destination. If voice call forwarding is active and multimedia call forwarding is not active then multimedia calls going off of the Avaya DEFINITY Server will be forwarded as voice only calls.

  The mm-cfwd button status lamp will be lit to indicate that multimedia call forwarding is activated. Pressing the mm-cfwd button when the lamp is lit will deactivate multimedia call forwarding.

  Note - pressing the mm-cfwd button is the same as dialing the regular call-fwd FAC followed by the mm-call button or FAC followed by the desired forwarded-to extension digits.

- **mm-multinbr** — The mm-multinbr call button is similar to the mm-call button. It allows origination of a multimedia call from any voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of address must be entered.

  Originating a multimedia call with the mm-multinbr button will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

ESM T.120 Server Administration

```
change ds1 1c09                                                    Page   1 of   1
                                DS1 CIRCUIT PACK

            Location: 01C09                          Name:
            Bit Rate: 2.048                   Line Coding: hdb3

      Signaling Mode: isdn-pri
             Connect: pbx                       Interface: network
                                         Country Protocol: 1

                                                     CRC? y
           Idle Code: 11111111
                              DCP/Analog Bearer Capability: 3.1kHz
   MMI Cabling Board: 01C10                   MMI Interface: ESM



      Slip Detection? n                Near-end CSU Type: other
```

Screen 21. DS1 screen (Page 1 of 1)

From the system administration terminal:

1. Enter **list configuration all,** and a list of the installed carriers, circuit packs, and ports appears.
2. Record the location (board number) of the MMI board cabled to the TN2207 slot and verify that all other required circuit packs are present.
3. Enter **add DS1 xxxxx,** (where xxxxx is the location of the TN2207 PRI circuit pack recorded in step 2), and the DS1 circuit pack administration screen appears.
4. Set the Name field to **ESM DS1**
5. Set the Bit Rate field to **2.048**

   The TN2207 DS1 must have a bit rate of 2.048, even if all other DS1 boards in the system are operating at 1.544. Verify the 24/32 channel switch on the circuit pack is in the 32 channel position.
6. Set the Line Coding field to **hdb3**
7. Set the Signaling Mode field to **isdn-pri**
8. Set the Connect field to **pbx**
9. Set the Interface field to **network**
10. Set the Country Protocol `field` to **1**
11. Set the CRC field to **y**
12. The Idle Code default is 11111111.

13. The DCP/Analog Bearer Capability default is 3.1 kHz.
14. Set the MMI Cabling Board field to **xxxxx** (where xxxxx is the location of the TN787 MMI circuit pack recorded in step 2). This must be the slot for port B of the Y-cable.
15. The MMI Interface field **ESM** appears.

```
change signaling-group 6                                        Page   1 of   5
                                SIGNALING GROUP

 Group Number: 6
                   Associated Signaling? y          Max number of NCA TSC: 0
                      Primary D-Channel: 01C0916     Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC:
       Trunk Group for Channel Selection:
          Supplementary Service Protocol: a
```

Screen 22. Signaling-Group screen (Page 1 of 5)

1. Enter **add signaling-group next** and the signaling-group screen appears.
2. Change Associated Signaling field to **y**.
3. Change Primary D-Channel Port field to **xxxx16** (where xxxx is the address of the TN2207 PRI circuit pack, for example: 1B0516).
4. The Max Number of NCA TSC default is 0.
5. The Max Number of GA TSC default is 0.
6. Trunk Group for NCA TSC ____ (leave blank)
7. Trunk Group for Channel Selection____ (leave blank)
8. Logoff the terminal and then log back on the terminal to view your changes.

Troubleshooting. To determine ESM link status enter the following commands from the system administration terminal:

1. **Status esm**
2. **Status signaling-group**
3. **List MMI**

NOTE:
When you move ESM circuit packs, you MUST remove the DS1 and signaling group translations. You cannot use the **change circuit pack** command.

When a vector is used to route video (56K/64K) calls to a hunt group comprised of data extensions, the vector ***must*** have the multimedia field set to **n**. This field causes multimedia calls routed through the vector to receive early answer treatment prior to processing the vector steps. This provides a talk path to the caller for announcements or immediate conversation with an agent and starts network billing for the incoming call when vector processing begins.

Understanding the multimedia complex

1-number access. 1-number access permits originating users to make voice or multimedia calls to a Basic multimedia complex by dialing the same number for either type of call. The number may be the voice station extension or the data module extension. If the incoming call is a voice call, Avaya MultiVantage directs it to the telephone. If the incoming call is 56K or 64K data call, Avaya MultiVantagerecognizes it as such and sends it to the multimedia endpoint. Likewise, if a voice call is addressed to the data extension, the system recognizes this and directs the call to the voice station.

Calls originating on the same switch as the Basic mode complex destination may always use 1-number access for voice or video. In order to take advantage of 1-number access for calls originating from a remote location, the incoming calls must arrive over ISDN-PRI trunks. If the system is setup with separate data non-PRI digital facilities multimedia calls must be made to the data extension.

AVD (Alternate voice/data) trunk groups cannot be used to provide 1-number access with MMCH. If the AVD trunk group has a BCC of 0, all calls arriving over the AVD trunk to the Basic mode complex will be assumed to be voice calls. If the AVD trunk group has a BCC of 1 or 4, all calls arriving over the AVD trunk to the Basic mode complex will be assumed to be multimedia calls.

Originating voice calls. All voice calls are originated at the voice station.

Originating multimedia calls. For a Basic mode complex, multimedia calls are normally originated at the user’s multimedia equipped PC. These multimedia calls use the associated station’s COR/COS.

The voice station of a Basic multimedia complex may also use the “mm-call” button or FAC, and the “mm-multinbr” button or FAC to originate multimedia calls. When these methods are used, a multimedia call is originated from the voice station. In order for the Basic multimedia complex to receive video, the user must make a call from the H.320 DVC system to the voice station of the complex or must make a multimedia call from the voice station to the H.320 DVC. This allows the station to spontaneously add themselves or other parties to a multimedia conference.

1. **H.320 DVC system GUI.** The normal way for a Basic multimedia complex endpoint to originate a multimedia call is through the vendor provided user interface. Generally, digits to dial are entered, speed is selected and the call originates from the DVC system. The voice station is not involved in such as origination.

   Any voice station may use the following mechanisms to originate a multimedia call from the voice station. For stations that are not part of a multimedia complex, video cannot be provided. For voice stations that are part of a Basic multimedia complex, video is not provided until a multimedia call is made from the complex’s H.320 DVC system to the voice station or a multimedia call is made from the voice station to the H.320 DVC system. Video is automatically included for Enhanced multimedia complexes.

2. **mm-call (Multimedia Call) button**. If the station has an mm-call button administered, the user goes off-hook and selects the mm-call button. The user may select the mm-call button and then go off-hook. If the user has a speakerphone on the station, the user may originate the call simply by selecting the mm-call button. The speakerphone will automatically be placed off-hook and dialtone will be heard. Upon selection of the mm-call button, the mm-call status lamp (green led) should become solid.

   The user now dials the destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, station busy indicators, etc. Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   For calls with a bandwidth of 2B, use of the mm-call button to originate will cause the same destination address to be used for both channels of the 2B call. See the section below on the mm-multinbr button/FAC for information on originating a 2B call where the destination has a different address for each B-channel.

   NOTE:
   The mm-call feature button is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a multimedia call.

3. **Multimedia Call feature Access Code**. For stations that do not have an administered mm-call button, the Multimedia call feature access code may be used instead. The user goes off-hook on the station, waits for dialtone, then dials the MM-call FAC, receives dialtone again and then dials the call normally. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, station busy indicators, etc.

Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

For calls with a bandwidth of 2B, use of the mm-call button to originate will cause the same destination address to be used for both channels of the 2B call. See the section below on the mm-multinbr button/FAC for information on originating a 2B call where the destination has a different address for each B-channel.

NOTE:
The mm-call feature access code is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a multimedia call.

4. **mm-multinbr (Multimedia Multi-number) button.** The mm-multinbr button is similar to the mm-call button. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

   The user goes off-hook and selects the mm-multinbr button. The user may select the mm-multinbr button and then go off-hook. If the user has a speakerphone on the station, the user may originate the call simply by selecting the mm-multinbr button. The speakerphone will automatically be placed off-hook and dialtone will be heard. Upon selection of the mm-multinbr button, the mm-multinbr and mm-call (if present) status lamp (green led) should light steadily. The user now dials the first destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. The system will provide dialtone after the first address has been completed. The user now dials the second destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. After the 2nd address has been collected the mm-multinbr status lamp will go off.

   Originating a multimedia call with the mm-multinbr button will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

NOTE:
The mm-multinbr feature button is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a dual address multimedia call.

5. **Multimedia Multi-number Call feature Access Code**. For stations that do not have an administered mm-multinbr button, the Multimedia Multi-number call feature access code may be used instead. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

   The user goes off-hook and dials the MM-multinbr feature access code. Upon dialing of the MM-multinbr FAC, the mm-call (if present) status lamp (green led) should become solid. The user now dials the first destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. The system will provide dialtone after the first address has been completed. The user now dials the second destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc.

   Originating a multimedia call with the MM-multinbr FAC will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   NOTE:
   The mm-multinbr FAC is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a dual address multimedia call.

6. **Multimedia parameter selection feature access code.** This FAC is used to originate a multimedia call that wishes to use a different bearer and bandwidth than the system default. For example, if the system has a default multimedia parameter of 2x64 and the user wishes to make a call to a destination that is known to only have 56K digital facilities, the MM parameter selection FAC can be used to select a bearer and bandwidth of 2x56 for this specific call.

The MM parameter selection FAC may be used in conjunction with the mm-multinbr button or FAC to make a single or dual address multimedia call at the desired bearer and bandwidth. The user goes off-hook and dials the MM-parameter selection feature access code. Dialtone is returned. The user enters a single digit, 1 or 2, where 1 = 2x64, 2 = 2x56. All other digits will produce reorder. Dialtone is returned. Upon dialing of the MM-parameter selection FAC, the mm-call (if present) status lamp (green led) should become solid. The user may indicate a dual-address call at this point with the mm-multinbr button or FAC. The user now dials one or two sets of destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc.

NOTE:
The mm-parameter selection FAC is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a dual address multimedia call.

7. Dialing sequences that include TACs, AAR, ARS, Authorization codes, CDR account codes, FRLs
    1. Single address with TAC
        - **Dial** mm-call button or FAC, **Hear** dialtone
        - **Dial** TAC, **Dial** destination digits
    2. Dual address with TAC
        - **Dial** mm-multinbr button or FAC, **Hear** dialtone
        - **Dial** TAC, **Dial** 1st dest. digits, **Hear** dialtone
        - **Dial** TAC, **Dial** 2nd dest. digits
    3. Single address with AAR/ARS
        - **Dial** mm-call button or FAC, **Hear** dialtone
        - **Dial** AAR/ARS, **Dial** destination digits
    4. Dual address with AAR/ARS
        - **Dial** mm-multinbr button or FAC, **Hear** dialtone
        - **Dial** AAR/ARS, **Dial** 1st dest. digits, **Hear** dialtone
        - **Dial** AAR/ARS, **Dial** 2nd dest. digits
    5. Single address with AAR/ARS and authorization code
        - **Dial** mm-call button or FAC, **Hear** dialtone
        - **Dial** AAR/ARS FAC, **Dial** destination digits, **Hear** stutter dialtone
        - **Dial** authorization code

6. Dual address with AAR/ARS and authorization code
   - **Dial** mm-multinbr button or FAC, **Hear** dialtone
   - **Dial** AAR/ARS FAC, **Dial** 1st dest. digits, **Hear** dialtone
   - **Dial** AAR/ARS FAC, **Dial** 2nd dest. digits, **Hear** stutter dialtone
   - **Dial** authorization code
7. Single address with TAC or AAR/ARS and CDR account code
   - **Dial** mm-call button or FAC, **Hear** dialtone
   - **Dial** CDR FAC, **Hear** dialtone
   - **Dial** CDR account code, **Hear** dialtone
   - **Dial** TAC or AAR/ARS, **Dial** destination digits
8. Dual address with TAC or AAR/ARS and CDR account code
   - **Dial** mm-multinbr button or FAC, **Hear** dialtone
   - **Dial** CDR FAC, **Hear** dialtone
   - **Dial** CDR account code, **Hear** dialtone
   - **Dial** TAC or AAR/ARS, **Dial** 1st dest. digits
   - **Dial** TAC or AAR/ARS, **Dial** 2nd dest. digits

Receiving voice calls. Any voice calls directed to the voice or data extension of a Basic multimedia complex will ring at the voice station.

Receiving multimedia calls. Any data calls directed to the voice or data extension of a Basic multimedia complex will ring at the multimedia equipped PC if it is available. You may answer the multimedia call at the PC and voice and video will connect to the PC. If the data endpoint is unavailable, the system verifies that the telephone of the complex is administered with the H.320 field set to **y**. If so, the system converts the call to voice and sends it to the telephone of the multimedia complex, where the call then alerts.

Hourglass Tone. When a voice station answers a converted multimedia call, the answering party may hear different things depending on the nature of the originator. If the origination is directly from an H.320 DVC system or if the originator is an Enhanced mode complex on a remote switch, an immediate audio path will not exist between the two parties. This is because the H.320 protocol must be established after the call is answered. It takes several seconds for the H.320 protocol to establish an audio path. During this interval the answering party will hear special ringback. When the audio path exists the special ringback will be removed and replaced with a short incoming call tone indicating that audio now exists. The combination of special ringback followed by incoming call tone is referred to as “**hourglass tone**.” Hourglass tone is an indication to the answering party that they should wait for the H.320 call to establish audio.

Early Answer. The answering party may administer their station to avoid hearing hourglass tone. With the station screen **Early Answer** field set to **y**, the system answers the incoming multimedia call on behalf of the station and establishes the H.320 protocol. After audio path has been established, the call will then alert at the voice station of the Basic complex destination. The station may then answer by going off-hook and will have immediate audio path. No hourglass tone will be heard by the answering party.

If the H.320 field is not set to **y** for the telephone of a Basic multimedia complex, H.320 calls alert at the multimedia endpoint until the caller drops. If an H.320 call is directed to a telephone with H.320 set to **n**, the system denies the call.

You can assign H.320 conversion to any voice station.

Authorization. Multimedia complexes require the same types of authorization (COR/COS) as standard telephones. If a call is addressed to the voice extension, the system checks the COR/COS of the telephone, whether the call is voice-only or multimedia. If a call is addressed to the data extension, the system checks the COR/COS of the data endpoint. If the call is subsequently redirected to the voice station, the system does a second COR/COS check for the authorization of the voice station. Calls originated from the PC use the COR/COS of the voice station.

Adjunct Switch Applications Interface . ASAI is not expected to support call-association for data calls. Therefore Avaya does not recommend that you use ASAI for multimedia.

Administered Connections. Basic Multimedia endpoints may serve as the origination point or destination of an administered connection.

Authorization and Barrier Codes. Basic Mode multimedia users or off-premises PC users may not be able to respond to prompts for authorization or barrier codes. Multimedia endpoints do not recognize the prompts.

An on-premises user might be able to use Remote Access and enter the entire digit string at once before launching the call, but it would be better to eliminate the need for such codes for multimedia users who need to call off premises.

Bridged Appearances. Voice users can bridge onto a call if the user has a bridged appearance of a voice member of the call.

Call redirection. Calls directed to either member of the Basic multimedia complex are subject to redirection (coverage, forwarding). Avaya MultiVantage converts calls to voice before sending them to coverage. Calls redirected through call forwarding maintain multimedia status if forwarded from the data endpoint.

Conferencing . A multimedia conference can consist of multimedia and voice-only conferees. All multimedia conferees are added to a multimedia conference by a voice-terminal user on the switch, who acts as the controller of the multimedia conference. When the controller is a Basic complex voice station, the controller must remain on the conference until all parties have joined. Once all endpoints are on the conference, the voice-terminal user may put the call on hold or drop, if the user wishes.

Video conferees can see only their local video and one other party. If more than two people are involved in a video conference, the person who is speaking is the one whose video appears to other conferees. The speaker’s video shows the previous speaker. This changes dynamically as the speaker changes.

Creating a multi-party video conference

All multimedia conferences must be controlled by a voice phone. Multimedia conferees may be added by calling the voice phone or by having the voice phone make a multimedia call to other DVC endpoints. The controller can then conference together individual parties to create a multimedia conference.

To set up a multimedia conference:

1. Determine who is going to be the conference controller.
2. At the appointed time, the conference controller calls his or her telephone from the multimedia endpoint by dialing the 1-number extension. Once this call is established, the controller conferences in other calls as if this were a voice conference. The controller continues to add conferees in this manner until all conferees have joined, or until the number of conferees reaches the administered limit.
3. The conference controller may also add voice or multimedia parties to the conference spontaneously. The controller hits CONFERENCE, makes a voice or multimedia call to a new party. To make a multimedia call, the controller must originate a call using the mm-call button or FAC or the mm-multinbr button or FAC. After the new party begins alerting, the controller may hit CONFERENCE to add the party to the existing conference on hold.

Coverage. Multimedia calls to a Basic mode complex are subject to the same coverage criteria as voice calls and follow the coverage path administered for the voice station of the Basic multimedia mode complex.

If a plain voice station or a Basic mode complex is the covering party, the answering voice station will receive audio only. If all voice stations in the coverage path have the station screen Early Answer field set to **n** and the originator of the multimedia call was not a local Enhanced mode complex, the answering station will hear hourglass tone.

If an Enhanced mode complex is the covering party, the answering voice station will receive voice and video. If all voice stations in the coverage path have the station screen Early Answer field set to **n** and the originator of the multimedia call was not a local Enhanced mode complex, the answering station will hear hourglass tone.

Coverage: Multimedia calls and off-net call coverage. If the principal station's coverage path include a remote coverage point, the multimedia call will cover off-switch as voice only. If the call is unanswered off-switch and proceeds to the next coverage point on-switch, the multimedia nature of the call is preserved.

Coverage: Multimedia calls and coverage to voice mail. Voice mail systems such as AUDIX are typically the last point in a coverage path and are usually implemented as a hunt group. In order to guarantee that the originator of an H.320 multimedia call hears the voice mail greeting, the hunt group that defines the list of voice mail ports should have the Early Answer field on the hunt group set to **y**. This field will have no effect on voice calls to the voice mail system.

Call Detail Recording. Each channel of a 2-channel call generates a separate CDR record.

Data Collaboration

Once you have established a multi-point video conference, multi-point T.120 data collaboration may be enabled for that call. This will allow all video parties on the current conference to collaborate.

T.120 Data conferencing is made possible through the Expansion Services Module (ESM) which is an adjunct to the Avaya MultiVantage. Up to six parties may participate in a single data conference, and up to 24 parties may use ESM facilities for data collaboration at any given time.

Adding data sharing to a video conference

1. Set up a multimedia conference.
2. Once a multimedia call is active, any voice station in the conference, can initiate data collaboration by pressing the mm-datacnf button. Or, to use the feature access code to initiate a data conference, press the Transfer button. A second line-appearance becomes active and you hear dial tone. Dial the multimedia data conference feature access code. Confirmation tone is heard and the system automatically reselects the held call appearance of the multimedia conference. The Avaya MultiVantage solution will select a data rate which is acceptable to all H.320 DVC systems in the current call.

   If the system does not have sufficient ESM resources available for all parties currently in the call, activation of T.120 data sharing will be denied. The mm-datacnf status lamp will flash denial or the mm-datacnf FAC will produce reorder.

3. Each H.320 DVC system in the conference call is joined to the data conference. On many DVC systems, the provided GUI may prompt the user with a dialog box, requesting the user to select a specific conference to join. With MMCH, there should only be one conference available to select.
4. The user must now use the PC's GUI to begin application sharing. The method for beginning application sharing or file transfer is different for each H.320 multimedia application. One of the H.320 DVC systems activates data sharing from the H.320 DVC vendor provided GUI. See your H.320 DVC system documentation for details.
5. The same H.320 DVC system as in step 4, opens an application, whiteboard, etc. to share and the image of the application is displayed on all H.320 DVC systems in the conference.

   For details on how multiple users may control the shared application, see the vendor provided documentation for your specific H.320 DVC system.
6. To end the data collaboration session and retain the voice/video conference, the station that selected the mm-datacnf button or FAC may press the mm-datacnf button or hit transfer and dial the mm-datacnf deactivation FAC.

   NOTE:
   As of this writing, many endpoints do not respond correctly to ending the data collaboration session and retaining voice/video. Some H.320 DVC systems drop the entire call. It is recommended that once T.120 data sharing has been enabled for a conference, that it remain active for the duration of the conference call. When all endpoints have dropped from the call, the T.120 resources will be released.

Joining a multimedia conference after T.120 data sharing has been enabled. If a multimedia conference with T.120 data sharing is already active and it is desired to conference in a new video endpoint, the new video endpoint can be conferenced into the existing call. The new endpoint will be allowed into the data conference if there exists sufficient ESM resources for the new endpoint. The new endpoint will get voice/video and data sharing if the new endpoint supports the MLP data rate chosen by the system when T.120 data collaboration was activated. If the endpoint does not support the pre-existing MLP data rate, the new endpoint will only receive voice and video.

Single switch data collaboration. When all parties involved in data collaboration conference are located on the same physical Avaya DEFINITY Server, there is no restriction on the type of user. The parties may be any combination of Enhanced multimedia complexes, Basic multimedia complexes or stand-alone H.320 DVC systems.

Multi- switch data collaboration. When all parties involved in data collaboration conference are **not** located on the same physical Avaya DEFINITY Server, the parties located on the Avaya DEFINITY Server hosting the data conference (i.e. the switch that activated mm-datacnf) may be any combination of Enhanced multimedia complexes, Basic multimedia complexes or stand-alone H.320 DVC systems. **All parties on remote switches must not be Enhanced multimedia complexes:** they must be Basic multimedia complexes or stand-alone H.320 DVC systems. Prior to originating or receiving a multimedia mode call, the mm-basic feature button or feature access code can be used to dynamically change an Enhanced mode complex into a Basic mode complex and back again.

Forwarding of voice/multimedia calls. In Basic mode you can forward calls from either the telephone or the multimedia endpoint. To forward a call from the multimedia endpoint:

1. At the PC’s multimedia application, enter the call-forwarding feature access code (FAC)
2. Enter the forward-to number in the dialed number field on the endpoint software
3. Click the Dial button (or equivalent)

NOTE:
The PC multimedia software will probably respond with a message that the call failed, since it does not recognize the FAC. In fact, Avaya MultiVantage *does receive* the message, and forwards all multimedia calls addressed to the 1-number.

If a call is forwarded from the telephone, the call converts to voice first. If using the multimedia endpoint to forward, the calls arrive at the forwarded-to extension as a data call. Such calls continue to ring until answered or abandoned, rather than follow a coverage path.

Users can forward calls from the multimedia endpoint using the call forward FAC. You can also assign a call-forward button at the voice station to forward calls for the data endpoint. If a Basic multimedia complex has console permissions, that user can forward calls for others by dialing the FAC, the data extension, and then the forwarded-to number.

Call Park. A voice-terminal user can park any active call, voice or multimedia, and unpark the call from another telephone. Users cannot park or unpark calls using multimedia endpoints.

Call Pickup. Members of a pickup group can answer an H.320 call using a telephone after the call has been converted to voice. This is true for standard or directed call pickup.

Consult. After a call is converted to voice, consult may be used when transferring or conferencing the call.

COR/COS. The Class of Restriction and Class of Service for H.320 calls originated from a 1-number complex are the same as those of the telephone in the complex.

Data Call Setup. Basic complex multimedia endpoints are BRI data endpoints, and may use data call-setup procedures as provided by the software vendor.

Data Hotline. If endpoint software allows users to select the dial function without entering a number, the endpoint can be used for hotline dialing.

Dial Access to Attendant. Access to Attendant is blocked for a data call from a Basic mode multimedia endpoint.

Data Trunk Groups. Data trunk groups may be used to carry H.320 calls of a fixed (administered) bearer capability.

Hold. The voice station and multimedia endpoint of a Basic complex are each independent devices with respect to call control. When a Basic multimedia complex voice station executes hold only the voice station is held. If the user has conferenced their multimedia endpoint into a multimedia conference, activating hold will ***not*** disconnect the multimedia endpoint from the conference, it will only disconnect the Basic multimedia complex voice station. Executing hold with an Enhanced mode complex will fully disconnect voice and video from the current active call.

Hunt Groups using Basic Mode complexes. Since Basic mode complexes may receive point to point multimedia calls at the DVC system and voice calls to the station simultaneously, the voice station extension may be placed in any normal voice hunt group or ACD skill and the data extension may be placed in a simple hunt group made up of only data extensions.

Basic mode complex data extensions or stand-alone data extensions may be used to create simple data hunt groups. Data extensions are not allowed in ACD hunt groups. It is recommended that you do not mix voice and data stations in a hunt group.

If you want multimedia calls to hunt to multimedia endpoints (i.e. 2B point to point data hunting), put the data extension in the hunt group. If you place the voice extension in a hunt group, only voice calls hunt to that extension. Multimedia calls to a hunt group with a Basic mode voice station as the hunt group member will ***not*** be offered to the DVC system of the Basic mode complex. If either the voice or data extension of a Basic mode complex is busy, the entire complex is considered busy for hunting purposes.

In order to guarantee that all members of a voice hunt group or skill can receive voice or multimedia calls, all members should have the H.320 field on the station screen set to **y**. Simple voice stations and Basic complex mode voice stations will receive voice only. Enhanced mode stations will receive voice and video.

The MM Early Answer field (on the Hunt Group screen) tells the system to answer the incoming multimedia call and establish audio before it reaches the first member of the hunt group. Thus, when the talk path is established, the caller is able to speak with an agent immediately. This is not necessary for hunt groups comprised of data extensions.

Hunting, Other considerations. Agents that are part of a Basic mode complex may dial a feature access code to remove themselves from availability (and to indicate that they are available again) from both the multimedia endpoint and the telephone independently. This allows the voice member or the data member to be individually made unavailable. To make the data extension unavailable, the agent must dial the FAC from the DVC system.

CMS measurements may indicate unusually slow ASA, because of the time required for the system to establish early-answer before offering the call to an agent.

Hunting Call association (routing). Typically incoming voice calls consist of 2 B-channel calls to the same address, to provide greater bandwidth and better video resolution. Avaya MultiVantage attempts to correctly pair up incoming calls and offer them as a unit to a single agent. MMCH uses call association to route both calls to the extension that answered the first call, regardless of how the call was routed internally.

Two 56K/64K data calls with the same calling party number to the same destination number are considered to be associated. The system makes every attempt to route both calls of a 2-channel call to the same answering party. If the first call terminates at a member of a hunt group, the second call does not have to hunt, but goes directly to the same member. In order for 2B multimedia calls to be correctly given to a single agent, incoming calls to the hunt group must have ANI information. The ANI information may be in the form of ISDN calling party number or DCS calling party number. Multimedia calls made on the same switch as the hunt group are easily associated. If multimedia calls into a hunt group have incorrect ANI information (i.e. all calls from switch X to switch Y include the LDN for switch X), then as the volume of calls increases, the number of mis-associated calls will increase. If multimedia calls into a hunt group have no ANI information, the switch will never associate pairs of calls and all calls will be treated independently and routed to separate agents. This is not a recommended configuration.

Hunting with Multimedia vectors. Very often, calls are routed to hunt groups or skills via a vector. The existing VDNs and vectors which exist for routing voice calls can be used to route multimedia calls.

In order to use a vector for multimedia calls that will terminate to voice stations, you must set the Multimedia field on the vector screen to **y**. This field has no effect on voice calls routing through the vector. This field will cause multimedia calls routed through the vector to receive early answer treatment prior to processing the vector steps. This provides a talk path to the caller for announcements or immediate conversation with an agent.

NOTE:
Vectors which have the Multimedia field set to y must eventually route to hunt groups, skills or numbers which are voice extensions. A vector with the multimedia field set to y should never be set up to route to a hunt group or number which is a data extension.

When a vector is used to route video (56K/64K) calls to a hunt group comprised of data extensions, the vector ***must*** have the multimedia field set to **n**.

Intercept Treatment. H.320 calls that receive intercept treatment are treated like other data calls. H.320 calls cannot be directed to an attendant for service because the attendant cannot have H.320 conversion service.

ISDN Trunk Groups. Avaya highly recommends that you use ISDN trunks for multimedia calls. ISDN PRI trunks allow complete 1-number access for an Enhanced multimedia complex. ANI provided over PRI trunks allows correct routing of multiple bearer channels to the correct destination device. ISDN also provides the bearer capability on a call by call basis which can be used to distinguish voice calls from multimedia calls.

Malicious Call Trace. If a malicious call terminates at a Basic multimedia complex endpoint, the user can dial the feature access code from the telephone to activate malicious call trace, followed by the extension of the multimedia endpoint. If the user does not dial the multimedia extension, MCT traces any call held on the telephone.

Message Waiting. Message Waiting indication is handled at the telephone. Because H.320 calls are converted to voice before going to coverage, all messages are voice only.

Night Service. Incoming Basic mode data calls follow established night-service processing for data calls.

Remote Access. The switch does not prevent Basic multimedia complexes from attempting to use remote access. However, these Basic mode endpoints will most likely not be able to dial the necessary codes.

Station Hunting . Basic mode data calls to endpoints that have an extension administered in the hunt-to-station field hunt based on established hunting criteria. The call is converted to voice before station hunting.

Tenant Partitioning. Permission to make multimedia calls or add parties of any type to a conference is subject to standard tenant-partitioning restrictions.

Terminating Extension Groups. Basic mode data calls to a TEG are converted to voice and can terminate only at a voice endpoint. Effectively, Avaya MultiVantage treats the multimedia-complex extension as a voice-only endpoint.

Telephone Display. Display information for calls to or from a Basic multimedia complex contains the 1-number.

Enhanced Mode MM complex

The Enhanced multimedia complex provides a much greater unified and integrated interface for control of voice and multimedia calls. The multifunction voice station is used to control all calls, whether voice or multimedia. The H.320 desktop video system is used to present the video stream, data stream and (optionally) audio stream to the user. The H.320 desktop video system is ***not*** used for call control. The Enhanced multimedia complex allows the multifunction voice station to handle voice or multimedia calls in an almost identical manner. Each call appearance on the voice station may represent a voice or multimedia call, allowing multiple voice or multimedia calls to be present simultaneously on the station. The user may manage the separate call appearances without regard to the voice or multimedia nature of the specific call. The standard HOLD/TRANSFER/CONFERENCE/DROP actions may be applied to any call without regard to the voice or multimedia nature of the call.

1-number access

1-number access permits originating users to make voice or multimedia calls to an Enhanced multimedia complex by dialing the same number for either type of call. The number may be the voice station extension or the data module extension. If the incoming call is a voice call, Avaya MultiVantage alerts the station of an incoming voice call. If the incoming call is 56K or 64K data call, Avaya MultiVantage recognizes it as a multimedia call and inserts resources to terminate the H.320 protocol and then alerts the voice station with a multimedia call.

Calls originating on the same switch as the Enhanced mode complex destination may always use 1-number access for voice or video. In order to take advantage of 1-number access for calls originating from a remote location, the incoming calls must arrive over ISDN-PRI trunks. If the system is setup with separate non-PRI digital facilities for data, multimedia calls must be made to the data extension of the Enhanced mode complex.

AVD (Alternate voice/data) trunk groups cannot be used to provide 1-number access with MMCH. If the AVD trunk group has a BCC of 0, all calls arriving over the AVD trunk to the Basic mode complex will be assumed to be voice calls. If the AVD trunk group has a BCC of 1 or 4, all calls arriving over the AVD trunk to the Basic mode complex will be assumed to be multimedia calls.

ORIGINATION

The basic call sequence from an Enhanced mode complex is to originate a multimedia call and alert the destination. When the destination answers the call, the originating station's H.320 desktop video system will be alerted (that is, called by the switch to establish the service link). If the H.320 desktop video system is not configured for auto-answer, the user must answer the H.320 calls via the DVC GUI. If the H.320 DVC is configured for auto-answer, no action is needed via the DVC GUI. **It is recommended, but not required, that Enhanced mode complexes place their desktop video system into an auto-answer mode of operation.** If the far-end is providing a video signal, the 2-way video will be observed. If the destination is not providing a video signal (call was answered by a simple voice station), then loopback video will be provided at the Enhanced mode complex originator. The audio signal will exist at the handset of the voice station. The audio signal may be moved to the H.320 DVC system via activation of a mm-pcaudio button on the voice station. See the section below on mm-pcaudio.

Hourglass tone. The originating party may hear different things when the incoming multimedia call is answered depending on the nature of the answering party. If the call is being answered directly by an H.320 DVC system or if the answering party is an Enhanced mode complex on a remote switch, an immediate audio path will not exist between the two parties. This is because the H.320 protocol must be established after the call is answered. It takes several seconds for the H.320 protocol to establish an audio path. During this interval the originating party will hear special ringback. When the audio path exists the special ringback will be removed and replaced with a short incoming call tone indicating that audio path now exists. The combination of special ringback followed by incoming call tone is referred to as “hourglass tone.” Hourglass tone is an indication to the originating party that they should wait for the H.320 call to establish audio.

- originating voice calls

Voice calls are originated from the voice station of an Enhanced mode complex in the normal manner as for any voice station.

- originating multimedia calls

Multimedia calls from an Enhanced multimedia complex are originated from the VOICE STATION, NOT the H.320 desktop video system. All multimedia originations require the user to indicate the multimedia nature of the call prior to providing any address digits. There are several different ways to originate a multimedia call from the voice station.

1. **mm-call (Multimedia Call) button.** If the station has an mm-call button administered, the user goes off-hook and selects the mm-call button. The user may select the mm-call button and then go off-hook. If the user has a speakerphone on the station, the user may originate the call simply by selecting the mm-call button. The speakerphone will automatically be placed off-hook and dialtone will be heard. Upon selection of the mm-call button, the mm-call status lamp (green LED) will light steadily, indicating a multimedia call.

   The user now dials the destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, station busy indicators, etc. Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   For calls with a bandwidth of 2B, use of the mm-call button to originate will cause the same destination address to be used for both channels of the 2B call. See the section below on the mm-multinbr button/FAC for information on originating a 2B call where the destination has a different address for each B-channel.

   NOTE:
   The mm-call feature button is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a multimedia call.

2. **Multimedia Call feature Access Code**. For stations that do not have an administered mm-call button, the Multimedia call feature access code may be used instead. The user goes off-hook on the station, waits for dialtone, then dials the MM-call FAC, receives dialtone again and then dials the call normally. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, station busy indicators, etc.

   Originating a multimedia call with the mm-call button will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   For calls with a bandwidth of 2B, use of the mm-call button to originate will cause the same destination address to be used for both channels of the 2B call. See the section below on the mm-multinbr button/FAC for information on originating a 2B call where the destination has a different address for each B-channel.

NOTE:
The mm-call feature access code is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a multimedia call.

3. **mm-multinbr (Multimedia Multi-number) button**. The mm-multinbr button is similar to the mm-call button. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

   The user goes off-hook and selects the mm-multinbr button. The user may select the mm-multinbr button and then go off-hook. If the user has a speakerphone on the station, the user may originate the call simply by selecting the mm-multinbr button. The speakerphone will automatically be placed off-hook and dialtone will be heard. Upon selection of the mm-multinbr button, the mm-multinbr and mm-call (if present) status lamp (green led) should become solid. The user now dials the first destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. The system will provide dialtone after the first address has been completed. The user now dials the second destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. After the 2nd address has been collected the mm-multinbr status lamp will go off.

   Originating a multimedia call with the mm-multinbr button will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

NOTE:
The mm-multinbr feature button is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a dual address multimedia call.

4. **Multimedia Multi-number Call feature Access Code.** For stations that do not have an administered mm-multinbr button, the Multimedia Multi-number call feature access code may be used instead. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. An example of this is Central Office provided ISDN-BRI. This type of BRI line is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

   The user goes off-hook and dials the MM-multinbr feature access code. Upon dialing of the MM-multinbr FAC, the mm-call (if present) status lamp (green led) should become solid. The user now dials the first destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc. The system will provide dialtone after the first address has been completed. The user now dials the second destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc.

   Originating a multimedia call with the MM-multinbr FAC will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen. This default parameter will be either 2x56 or 2x64. The bearer capability of the multimedia calls will either be 56K or 64K and the bandwidth will be 2B channels.

   NOTE:
   The mm-multinbr FAC is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a dual address multimedia call.

5. **Multimedia parameter selection feature access code.** This FAC is used to originate a multimedia call that wishes to use a different bearer and bandwidth than the system default. For example, if the system has a default multimedia parameter of 2x64 and the user wishes to make a call to a destination that is known to only have 56K digital facilities, the MM parameter selection FAC can be used to select a bearer and bandwidth of 2x56 for this specific call.

   The MM parameter selection FAC may be used in conjunction with the mm-multinbr button or FAC to make a single or dual address multimedia call at the desired bearer and bandwidth. The user goes off-hook and dials the MM-parameter selection feature access code. Dialtone is returned. The user enters a single digit, 1 or 2, where 1 = 2x64, 2 = 2x56. All other digits will produce reorder. Dialtone is returned. Upon dialing of the MM-parameter selection FAC, the mm-call (if present) status lamp (green led) should become solid. The user may indicate a dual-address call at this point with the mm-multinbr button or FAC. The user now dials one or two sets of destination address digits. The destination address may be provided by dialing digits, using abbreviated dial entries, last number dialed, etc.

NOTE:
The mm-parameter selection FAC is generally used by stations that are part of an Enhanced multimedia complex, but may be used by any station to originate a dual address multimedia call.

6. Dialing sequences that include TACs, AAR, ARS, Authorization codes, CDR account codes, FRLS
   1. Single address with TAC
      - **Dial** mm-call button or FAC, **Hear** dialtone
      - **Dial** TAC, **Dial** destination digits
   2. Dual address with TAC
      - **Dial** mm-multinbr button or FAC, **Hear** dialtone
      - **Dial** TAC, **Dial** 1st dest. digits, **Hear** dialtone
      - **Dial** TAC, **Dial** 2nd dest. digits
   3. Single address with AAR/ARS
      - **Dial** mm-call button or FAC, **Hear** dialtone
      - **Dial** AAR/ARS, **Dial** destination digits
   4. Dual address with AAR/ARS
      - **Dial** mm-multinbr button or FAC, **Hear** dialtone
      - **Dial** AAR/ARS, **Dial** 1st dest. digits, **Hear** dialtone
      - **Dial** AAR/ARS, **Dial** 2nd dest. digits
   5. Single address with AAR/ARS and authorization code
      - **Dial** mm-call button or FAC, **Hear** dialtone
      - **Dial** AAR/ARS FAC, **Dial** destination digits, **Hear** stutter dialtone
      - **Dial** authorization code
   6. Dual address with AAR/ARS and authorization code
      - **Dial** mm-multinbr button or FAC, **Hear** dialtone
      - **Dial** AAR/ARS, **Dial** 1st dest. digits, **Hear** dialtone
      - **Dial** AAR/ARS, **Dial** 2nd dest. digits, **Hear** stutter dialtone
      - **Dial** authorization code

7. Single address with TAC or AAR/ARS and CDR account code
   - **Dial** mm-call button or FAC, **Hear** dialtone
   - **Dial** CDR FAC, **Hear** dialtone
   - **Dial** CDR account code, **Hear** dialtone
   - **Dial** TAC or AAR/ARS, **Dial** destination digits
8. Dual address with TAC or AAR/ARS and CDR account code
   - **Dial** mm-multinbr button or FAC, **Hear** dialtone
   - **Dial** CDR FAC, **Hear** dialtone
   - **Dial** CDR account code, **Hear** dialtone
   - **Dial** TAC or AAR/ARS, **Dial** 1st dest. digits
   - **Dial** TAC or AAR/ARS, **Dial** 2nd dest. digits

Answering

The user actions required to answer voice or multimedia calls at an Enhanced multimedia complex are identical if the H.320 DVC system is configured for auto-answer. If the H.320 DVC system is not configured for auto-answer an additional step is required. See answering multimedia calls below. **It is recommended, but not required, that Enhanced mode complexes place their desktop video system into an auto-answer mode of operation.**

Answering voice calls

Incoming voice calls will alert at the voice station of the Enhanced multimedia complex in the normal manner. Standard alerting and call appearance flashing will occur. They are answered in the normal manner by selecting the alerting call appearance and going off-hook on the voice station.

Answering multimedia calls

Incoming multimedia calls will alert at the voice station of the Enhanced multimedia complex in the same manner as voice calls with one addition. If the alerting station has an administered mm-call button and the alerting call appearance is the selected call appearance (i.e. the red led is lit, on the alerting call appearance), then the mm-call button status lamp will go on indicating that the call on the selected call appearance is a multimedia call.

The incoming multimedia call is answered in the normal manner by selecting the alerting call appearance and going off-hook on the voice station. If the H.320 DVC system for the answering party is configured for auto-answer, no other action is needed to complete the multimedia call. If the H.320 DVC system for the answering party is not configured for auto-answer, the H.320 DVC system will alert and must also be answered by the user. **It is recommended, but not required, that Enhanced mode complexes place their desktop video system into an auto-answer mode of operation.**

If the originating party is providing a video signal, then a complete 2-way multimedia call will exist. If the originating party is not providing a video signal, the answering party will receive loopback video. The audio signal will exist at the handset of the voice station. The audio signal may be moved to the H.320 DVC system via activation of a **mm-pcaudio** button on the voice station.

Hourglass Tone. The answering party may hear different things when the incoming multimedia call is answered depending on the nature of the originator. If the origination is directly from an H.320 DVC system or if the originator is an Enhanced mode complex on a remote switch, an immediate audio path will not exist between the two parties. This is because the H.320 protocol must be established after the call is answered. It takes several seconds for the H.320 protocol to establish an audio path. During this interval the answering party will hear special ringback. When the audio path exists the special ringback will be removed and replaced with a short “incoming call tone” indicating that audio now exists. The combination of special ringback followed by incoming call tone is referred to as **“hourglass tone.**” Hourglass tone is an indication to the answering party that they should wait for the H.320 call to establish audio.

Early Answer. The answering party may administer their station in such a way as to avoid hearing hourglass tone. If the station screen has set the **Early Answer** field to **y**, then the system will answer the incoming multimedia call on behalf of the station and proceed to establish the H.320 protocol. After audio path has been established, the call will then alert at the voice station of the Enhanced mode complex destination. The station may then answer by going off-hook and will have immediate audio path. No hourglass tone will be heard by the answering party.

- multiple call appearance operation

With an Enhanced mode complex all calls to or from the complex are controlled via the voice station. Each voice or multimedia call has its own call appearance which may be selected without regard for the nature of the call using the specific call appearance. This allows a multifunction station to control multiple voice or multimedia calls in exactly the same way they would control multiple voice calls.

As an example, a user may originate a simple voice call on the first call appearance. A multimedia call may then arrive on the second call appearance. The user activates HOLD on the first call appearance and selects the second call appearance to answer the multimedia call. The user may then activate HOLD on the second call appearance and reselect the first call appearance or select a third call appearance and originate another call.

- creating a multi-party video conference

An Enhanced multimedia complex can create a spontaneous video conference in the same way that a spontaneous voice conference is created. Given an active call, the user activates the CONFERENCE button. This puts the current call on HOLD and activates a new call appearance. The user makes a multimedia call according to the instructions for originating a multimedia call and then selects CONFERENCE to combine or merge the two call appearances. This results in a 3-way conference.

If all three parties are video equipped, then a 3-way video conference results. Conference members see the current speaker on video. The current speaker sees the last speaker on video. If one of the parties is not video equipped, then a 3-way audio conference exists and the two video equipped parties have 2-way video. The CONFERENCE action may be repeated until 6 parties have been conferenced together. The 6 parties may be any mix of voice or video, local or remote parties.

The following steps create a multi-party voice/video conference:

1. Enhanced mode complex station A originates a multimedia call to, or receives a multimedia call from, party B. Station A and party B have 2-way voice and video.
2. Station A, activates CONFERENCE.
3. Station A originates a multimedia call (i.e. uses the mm-call button/FAC/etc.) and dials the party to be added, Enhanced multimedia complex C.
4. Party C, answers the call from station A.
5. Station A selects CONFERENCE to complete the 3-way conference. Parties A,B and C will be in a 3-way voice/video conference.

NOTE:
If party C is **another Enhanced mode complex on the same switch as station A**, station A does not need to indicate a multimedia call prior to dialing the new party in step 3. While A consults with C, the call will be audio only. When A completes the conference in step 5, party C’s video will be activated.

A multi-party video conference uses voice activated switching to determine which parties are seen. The current speaker is seen by all other parties. The current speaker sees the previous speaker.

Additional voice or video parties may be added by repeating these steps.

Data Collaboration

Once you have established a multi-point video conference, multi-point T.120 data collaboration may be enabled for that call. This will allow all video parties on the current conference to collaborate.

T.120 Data conferencing is made possible through the Expansion Services Module (ESM) which is an adjunct to the Avaya DEFINITY Server. Up to six parties may participate in a single data conference, and up to 24 parties may use ESM facilities for data collaboration at any given time.

The following steps add data sharing to a video conference:

1. Set up a multimedia conference.
2. Once a multimedia call is active, any member can initiate data collaboration by pressing the mm-datacnf button. Or, to use the feature access code to initiate a data conference, press the Transfer button. A second line-appearance becomes active and you hear dial tone. Dial the multimedia data conference feature access code. Confirmation tone is heard and the system automatically reselects the held call appearance of the multimedia conference. Avaya MultiVantage will select an MLP data rate which is acceptable to all H.320 DVC systems in the current call.

   If the system does not have sufficient ESM resources available for all parties currently in the call, activation of T.120 data sharing will be denied. The mm-datacnf status lamp will flash denial or the mm-datacnf FAC will produce reorder.
3. Each H.320 DVC system in the conference call is joined to the data conference. On many DVC systems, the provided GUI may prompt the user with a dialog box, requesting the user to select a specific conference to join. With MMCH, there should only be one conference available to select.
4. The user must now use the PC’s GUI to begin application sharing. The method for beginning application sharing or file transfer is different for each H.320 multimedia application. One of the H.320 DVC systems activates data sharing from the H.320 DVC vendor provided GUI. See your H.320 DVC system documentation for details.
5. The same H.320 DVC system as in step 4, opens an application, whiteboard, etc. to share and the image of the application is displayed on all H.320 DVC systems in the conference.

   For details on how multiple users may control the shared application, see the vendor provided documentation for your specific H.320 DVC system.

6. To end the data collaboration session and retain the voice/video conference, the station that selected the mm-datacnf button or FAC may press the mm-datacnf button or hit transfer and dial the mm-datacnf deactivation FAC.

 NOTE:
 Currently, many endpoints do not respond correctly to ending the data collaboration session and retaining voice/video. Some H.320 DVC systems drop the entire call. It is recommended that once T.120 data sharing has been enabled for a conference, that it remain active for the duration of the conference call. When all endpoints have dropped from the call, the T.120 resources will be released.

Joining a multimedia conference after T.120 data sharing has been enabled.

If a multimedia conference with T.120 data sharing is already active and it is desired to conference in a new video endpoint, the new video endpoint can be conferenced into the existing call. The new endpoint will be allowed into the data conference if there exists sufficient ESM resources for the new endpoint. The new endpoint will get voice/video and data sharing if the new endpoint supports the data rate chosen by the system when T.120 data collaboration was activated. If the endpoint does not support the pre-existing data rate, the new endpoint will only receive voice and video.

Activating HOLD while on a T.120 data collaboration conference.

If an Enhanced multimedia complex is active on a multimedia call and the call has activated T.120 data collaboration, the user should be receiving voice/video and data. If the station places this existing call on hold, audio and video will be disconnected for the current call. The data collaboration portion of the call will remain intact and unaffected. While this T.120 data conference is on hold, the user will only be allowed to receive audio on all other call appearances. Thus a user is limited to one call appearance that has T.120 data collaboration active.

Single switch data collaboration. When all parties involved in data collaboration conference are located on the same physical Avaya DEFINITY Server, there is no restriction on the type of user. The parties may be any combination of Enhanced multimedia complexes, Basic multimedia complexes or stand-alone H.320 DVC systems.

Multi- switch data collaboration. When all parties involved in data collaboration conference are *not* located on the same physical Avaya DEFINITY Server, the parties located on the Avaya DEFINITY Server hosting the data conference (i.e. the switch that activated mm-datacnf) may be any combination of Enhanced multimedia complexes, Basic multimedia complexes or stand-alone H.320 DVC systems.

NOTE:
All parties on remote switches must not be Enhanced multimedia complexes. They must be Basic multimedia complexes or stand-alone H.320 DVC systems.

Prior to originating or receiving a multimedia mode call, the mm-basic feature button or feature access code can be used to dynamically change an Enhanced mode complex into a Basic mode complex and back again.

Voice station audio vs. H.320 DVC system audio

When an Enhanced mode complex originates or receives a voice or multimedia call, the call is originated with the station handset or answered with the station handset. The audio path will be through the handset. If the user's H.320 DVC system has speakers and a microphone, the user may wish to use the H.320 DVC system for audio in much the same manner as a built-in or separate telephone speakerphone. The user can move the station's audio to the H.320 DVC system by selecting an **mm-pcaudio** feature button on the voice station. There is no feature access code for this function.

The mm-pcaudio feature button works very much like a speakerphone on/off button. If the station is off-hook and selects mm-pcaudio, audio is directed to the PC DVC system. The switch-hook may be placed on-hook. If the handset is taken off-hook, the audio moves back to the handset. If the mm-pcaudio button is selected while audio is already on the DVC system and the handset is on-hook, this acts as a speakerphone off action and disconnects the current call.

The mm-pcaudio feature button may be used for voice as well as multimedia calls. If the mm-pcaudio feature button is selected while on a voice only call, the DVC system is alerted and brought into the call. No video will be transmitted or displayed. Audio will be directed through the PC DVC system.

Switching between Basic and Enhanced modes

There may exist occasions when an Enhanced mode complex needs to switch to Basic mode operation temporarily. One example is when a user wishes to make a direct point to point multimedia call originated directly from the H.320 DVC. Basic mode operation allows this functionality at the expense of losing multimedia call handling capabilities (i.e. hold/xfer/conf). To switch from Enhanced mode to Basic mode, the station may either select a mm-basic feature button or dial the mm-basic feature access code. Both of these actions are valid only if the Enhanced mode station has no multimedia calls active.

When in Basic mode, the status lamp for the mm-basic button, if present, will be on solid. The mm-basic feature button acts as a toggle. If the status lamp is on, when the button is selected, the lamp will go off and the station will return to Enhanced mode. The mm-enhanced feature access code will set the state of the station back to Enhanced. Switching to Enhanced mode is only valid if the associated H.320 DVC system is idle.

NOTE:
Toggling between Basic and Enhanced mode changes the station's administered Multimedia mode. When in Basic mode this field on the station screen will show `basic`. When in Enhanced mode this field on the station screen will show `enhanced`. The current station Multimedia mode will be saved to translation when a **save translation** command is executed.

Forwarding of voice and multimedia calls

The Enhanced multimedia mode complex voice station may use the existing standard call forwarding mechanisms to activate forwarding for voice calls. If the forwarding destination is on the same switch then this will also forward multimedia calls as multimedia calls to the destination. If the forwarding destination is off switch, multimedia calls will forward off switch as voice only calls. This is appropriate when the user will be at a location that is not able to receive multimedia calls.

To forward multimedia calls off switch as multimedia calls, the user must activate multimedia call forwarding. This may be done with an mm-cfwd button or feature access code. The user may also activate standard voice call forwarding and select the mm-call button prior to entering the forwarding address.

Coverage

Multimedia calls to an Enhanced mode complex are subject to the same coverage criteria as voice calls and follow the coverage path administered for the voice station of the Enhanced multimedia mode complex.

If a plain voice station or a Basic mode complex is the covering party, the answering voice station will receive audio only. If all voice stations in the coverage path have the station screen Early Answer field set to **n** and the originator of the multimedia call was not a local Enhanced mode complex, the answering station will hear hourglass tone.

If an Enhanced mode complex is the covering party, the answering voice station will receive voice and video. If all voice stations in the coverage path have the station screen Early Answer field set to **n** and the originator of the multimedia call was not a local Enhanced mode complex, the answering station will hear hourglass tone.

Multimedia calls and off-net call coverage. If the principal station's coverage path include a remote coverage point, the multimedia call will cover off-switch as voice only. If the call is unanswered off-switch and proceeds to the next coverage point on-switch, the multimedia nature of the call is preserved.

Multimedia calls and coverage to voice mail. Voice mail systems such as AUDIX are typically the last point in a coverage path and are usually implemented as a hunt group. In order to guarantee that the originator of an H.320 multimedia call hears the voice mail greeting, the hunt group that defines the list of voice mail ports should have the Early Answer field on the hunt group set to **y**. This field will have no effect on voice calls to the voice mail system.

Hunt Groups using Enhanced Mode Complexes

When creating hunt groups with Enhanced multimedia mode complexes, only the station extension should ever be entered as a hunt group member. Any hunt group or ACD skill can include the voice station of an Enhanced multimedia complex as a member. The data extension of an Enhanced mode complex should never be entered as any hunt group member. A hunt group or skill may have a mix of members that are stand-alone stations and Enhanced mode complex stations. In order to guarantee that all members of the hunt group or skill can receive voice or multimedia calls, all members should have the H.320 field on the station screen set to **y**. Simple voice stations will receive voice only. Enhanced mode stations will receive voice and video.

The MM Early Answer field (on the Hunt Group screen) tells the system to answer an incoming multimedia call and establish audio before it reaches the first member of the hunt group. Thus, when the talk path is established, the caller is able to speak with an agent immediately.

Other considerations. CMS measurements may indicate unusually slow ASA, because of the time required for the system to establish early-answer before offering the call to an agent.

Call association (routing). Typically incoming voice calls consist of 2 B-channel calls to the same address, to provide greater bandwidth and better video resolution. Avaya MultiVantage attempts to correctly pair up incoming calls and offer them as a unit to a single agent. MMCH uses call association to route both calls to the extension that answered the first call, regardless of how the call was routed internally.

Two 56K/64K data calls with the same calling party number to the same destination number are considered to be associated. The system makes every attempt to route both calls of a 2-channel call to the same answering party. If the first call terminates at a member of a hunt group, the second call does not have to hunt, but goes directly to the same member.

In order for 2B multimedia calls to be correctly given to a single agent, incoming calls to the hunt group must have ANI information. The ANI information may be in the form of ISDN calling party number or DCS calling party number. Multimedia calls made on the same switch as the hunt group are easily associated. If multimedia calls into a hunt group have insufficient ANI information (i.e. all calls from switch X to switch Y include the LDN for switch X), then as the volume of calls increases the number of mis-associated calls will increase. If multimedia calls into a hunt group have no ANI information, the switch will never associate pairs of calls and all calls will be treated independently and routed to separate agents. This is not a recommended configuration.

Multimedia vectors. Very often, calls are routed to hunt groups or skills via a vector. The existing VDNs and vectors which exist for routing voice calls can be used to route multimedia calls.

In order to use a vector for multimedia calls, you must set the Multimedia field on the vector screen to **y**. This field has no effect on voice calls routing through the vector. This field will cause multimedia calls routed through the vector to receive early answer treatment prior to processing the vector steps. This provides a talk path to the caller for announcements or immediate conversation with an agent.

> **NOTE:**
> Vectors which have the Multimedia field set must eventually route to hunt groups, skills or numbers which are voice extensions. A vector with the multimedia field set to y should never be set up to route to a hunt group or number which is a data extension.

Interactions

Interactions are listed here only if the operation is different from standard.

- Administered Connections

  An Enhanced multimedia complex voice station may serve as the origination point or destination of an administered connection. If the Multimedia call feature access code is included in the administration of the administered connection, this will result in a video AC.

  An Enhanced multimedia complex H.320 DVC system may not serve as the origination point of an administered connection.

- X-porting

  You cannot use X in the port field when administering a data module or the data endpoint in a multimedia complex. However, you can use this to administer the telephone.

- Bridged Appearances

  Enhanced multimedia complex voice station users can bridge onto a call if the user has a bridged appearance. If the bridged appearance is for a multimedia call, selecting the bridged appearance will result in a multimedia call.

- Call Detail Recording

  Each channel of a 2-channel multimedia call generates a separate CDR record that is tagged as data.

- Call forwarding

  Users cannot forward calls from a multimedia complex using multi-number dialing, either by mm-multnmbr button or feature access code.

- Call Park

  Any station can park a multimedia call, and unpark the call from another telephone. If a multimedia call is unparked by an Enhanced mode complex station, a multimedia call will result. Users cannot park or unpark calls using multimedia endpoints.

- Call Pickup

  Any member of a pickup group can answer a multimedia call after the call has begun alerting at a station call appearance. If the station picking up the call is an Enhanced mode complex station and the call is multimedia, a multimedia call will result. This is true for standard or directed call pickup.

- Consult

  After a multimedia call has been answered, consult may be used when transferring or conferencing the call.

- COR/COS

  The Class of Restriction and Class of Service for a multimedia call originated from an Enhanced multimedia complex are those of the voice station in the complex.

- Data Call Setup

  An Enhanced mode multimedia H.320 DVC system may not originate calls from the DVC system. All calls, both voice or video are originated from the voice station.

- Data Hotline

  An Enhanced multimedia complex H.320 DVC endpoint may not be used to originate a call for hotline dialing. In order to setup a video hotline function with an Enhanced mode complex, the hotline number administered for the voice station should include the Multimedia call feature access code.

- Data Trunk Groups

  Data trunk groups may be used to carry H.320 calls of a fixed (administered) bearer capability.

- ISDN Trunk Groups

  Avaya highly recommends that you use ISDN trunks for multimedia calls. ISDN PRI trunks allow complete 1-number access for an Enhanced multimedia complex. ANI provided over PRI trunks allows correct routing of multiple bearer channels to the correct destination device. ISDN also provides the bearer capability on a call by call basis that can be used to distinguish voice calls from multimedia calls.

- Night Service

  Incoming H.320 calls follow established night-service processing for data calls.

- Remote Access

  The switch does not prevent Enhanced multimedia complexes from attempting to use remote access. However, these endpoints will most likely not be able to dial the necessary codes.

- Station Hunting

  Multimedia calls to Enhanced mode complex voice stations that have an extension administered in the hunt-to-station field hunt based on established hunting criteria. If the hunt-to-station is also an Enhanced mode complex station, a multimedia call will result when the call is answered.

- Terminating Extension Groups

  A multimedia call to a TEG may be answered by any member of the TEG. If the member answering the call is an Enhanced mode complex station, a multimedia call will result.

- Telephone Display

  Display information for calls to or from an Enhanced multimedia complex contains the display information associated with the voice station.

Troubleshooting

If one channel of a 2 B-channel call goes down, your choices are to continue with reduced transmission quality, or to hang up the call and start over. It is not possible to re-establish the second channel while the call is still active.

If you cannot share data with others, it may be that both parties do not have the same endpoint software. This is true for some data collaboration, but most whiteboard and file transfer software implementations are compatible.

Monitoring MMCH

This section briefly discusses some of the commands you can use to monitor multimedia complexes and conferences. The Maintenance manual for your system discusses these commands and their output in detail.

Action	Objects	Qualifier
display	station data module	xxxxx (extension) xxxxx (extension)
list	mmi	
	measurements	multimedia-interface voice-conditioner esm
	multimedia	endpoints [’print’ or ’schedule’] h.320-stations [’print’ or ’schedule’]
status	attendant conference conference conference data module station trunk	xxxx (console number) all xxx (conference ID) xxx (conference ID) endpoint (endpoint ID) xxxxx (extension) xxxxx (extension) (group number or group number/member number)
	esm	

Status commands

The **status** commands for data module, station, trunk, and attendant provide the conference ID and endpoint ID for any of these involved in an active multimedia conference.

```
status station 1002
                           GENERAL STATUS

              Type: 7405D                 Service State: in-service/on-hook
         Extension: 1002               Maintenance Busy? no
              Port: 01C0702               SAC Activated? no
       Call Parked? no                 User Cntrl Restr: none
  Ring Cut Off Act? no                Group Cntrl Restr: none
Active Coverage Option: 1           CF Destination Ext:
                                       MM Conference ID:
                                         MM Endpoint ID:

   Message Waiting:
   Connected Ports:

             ACD STATUS                        HOSPITALITY STATUS
    Agent Logged In   Work Mode             AWU Call At:
                                               User DND: not activated
                                              Group DND: not activated
                                            Room Status: non-guest room
       On ACD Call? no
```

Screen 23. Status Station 1002 — General Status screen

The following fields specific to multimedia appear on the status station, attendant, data module and trunk screens.

- **MM Conference ID** — This field appears only if the station is active on a multimedia conference. It displays the ID for the conference. Enter this number with the status conference command to get more information about this conference.
- **MM Endpoint ID** — This field appears only if the station is active on a multimedia conference. It displays the endpoint ID for the station. Enter this number with the status conference endpoint command to learn more about this endpoint's involvement in the conference.

List commands

The **list multimedia endpoints** command shows you all the multimedia data modules that exist in your system, and their associated telephones, if any. The **list multimedia H.320-stations** command shows you all the stations that are administered for H.320 conversion.

```
             MULTIMEDIA ENDPOINTS

Data Ext     MM Complex Voice Ext     H.320 Conversion?

100                  87654                    y
1321
15683                738                      n
```

Screen 24. List Multimedia Endpoints screen

```
                    MULTIMEDIA H.320-STATIONS

        Station Ext             MM Data Ext

           100                     87654
           1321
           15683                   738
```

Screen 25. List Multimedia H.320-Stations screen

Considerations

Each channel of a 2-channel BRI call takes one port on an MMI circuit pack. This alone limits the number of multimedia calls your system can handle. In addition, each conference takes one port on a voice-conditioner circuit pack.

Also, there is a limit to the number of conversion calls that the system can handle simultaneously. If you experience traffic problems after installing multimedia, you may want to lower the number of stations that use H.320 conversion.

Setting up telecommuting

10

Configuring Avaya MultiVantage for telecommuting

Telecommuting emphasizes the ability to perform telephony activities while remote from the Avaya MultiVantage solution. It is a combination of four features which permit you to remotely perform changes to your station's Coverage and Call Forwarding.

NOTE:
If you are operating in a DCS environment, you need to assign a different telecommuting-access extension to each switch and tell your users which extension they should use. A user can set up call coverage from any of the DCS nodes, but needs to dial the telecommuting-access extension of the node on which their station is defined before using the feature access code.

You can also set up telecommuting with an IP (internet protocol) phone. See ''Adding an IP Softphone'' for more information.

- Coverage of Calls Redirected Off Net (CCRON) allows you to redirect calls off-net onto the public network and bring back unanswered calls for further coverage.

  NOTE:
  If a call covers or forwards off net and an answering machine answers the call or it is directed to a cellular phone and a cellular announcement is heard, the switch views this call as an answered call. The switch does not bring the call back to the switch for further routing.

- The Extended User Administration of Redirected Calls feature allows you to change the direction of calls to your station. This activates the capability to have 2 coverage path options. These 2 path options can be specified on the station screen; however, unless the Can Change Coverage field is set to **y** on the Class of Restriction screen the second path option cannot be populated.
- The Personal Station Access feature gives you an extension number, a Merge feature access code, and a personalized security code and tells you which office phone you can use. This allows you to take your phone, as long as the phones are the same type, anywhere on the same switch.
- The Answer Supervision feature provides supervision of a call directed out of the switch either by coverage or forwarding and determines whether the switch should bring the call control back.

Before you start

Ensure you have the following equipment:

- Tone Clock — Tone Detector and Call Classifier
- Call Classifier — Detector
- 1264-TMx software
- MultiVantage extender — switch module or standalone rack mount (Digital Communications Protocol (DCP) or Integrated Services Digital Network (ISDN))

  For more information about this equipment, refer to the *Avaya MultiVantage™ Hardware Solutions Guide*.

Verify the following fields on the System-Parameters Customer-Options screen are set to y.

- Cvg Of Calls Redirected Off-Net
- Extended Cvg/Fwd Admin
- Personal Station Access
- Terminal Translation Initialization (TTI)

If neither MultiVantage extender nor the System Parameters Customer Options fields are configured, contact your Avaya representative.

Verify the telecommuting access extension is a direct inward dialing (DID) or a central office (CO) trunk destination for off-premises features to work.

Configure terminal translation initialization (TTI) for personal station access (PSA). For information about configuring TTI, refer to ''Setting up Personal Station Access'' on page 354.

Configure Security Violation Notification for Station Security Codes. For information about Security Violation Notification, refer to ''Setting up security violations notification'' on page 395.

Instructions

In our example, we set up the telecommuting extension and enable coverage of calls redirected off-net.

To configure Avaya MultiVantage for telecommuting:

1. Type **change telecommuting-access** and press RETURN.

   The Telecommuting Access screen appears.

```
                    TELECOMMUTING ACCESS

          Telecommuting Access Extension: 1234
```

2. In the Telecommuting Access Extension field, type **1234** and press ENTER.

   This is the extension you are configuring for telecommuting.

3. Type **change system-parameters coverage** and press RETURN.

   The System Parameters Call Coverage / Call Forwarding screen appears.

4. In the Coverage Of Calls Redirected Off-Net Enabled field, type **y** and press ENTER.

Related topics

Refer to ''Telecommuting Access'' on page 1148 for information about and field descriptions on the Telecommuting Access screen.

Setting up Personal Station Access

Personal Station Access (PSA) allows you to associate the preferences and permissions assigned to your own extension with any other compatible phone. When you request a PSA associate, the system automatically dissociates another extension from the phone.

Preferences and permissions include the definition of terminal buttons, abbreviated dial lists, and class of service (COS) and class of restriction (COR) permissions assigned to your station. Extensions without a COS, such as Expert Agent Selection (EAS) agents or hunt groups, cannot use PSA.

PSA requires you to enter a security code and can be used on-site or off-site. Invalid attempts to associate a phone generate referral calls and are recorded by Security Violation Notification, if that feature is enabled. If you interrupt the PSA dialing sequence by pressing the release button or by hanging up, the system does not log the action as an invalid attempt.

The disassociate function within PSA allows you to restrict the features available to a phone. When a phone has been dissociated using PSA, it can be used only to call an attendant, or to accept a Terminal Translation Initialization (TTI) or PSA request. You can enable a dissociated set to make other calls by assigning a special class of restriction.

When a call that goes to coverage from a PSA-disassociated extension, the switch sends a message to the coverage point indicating that the call was not answered. If the coverage point is a display phone, the display shows “da” for “don't answer.” If the coverage point is a voice messaging system, the VM system receives an indication from the switch that this call was not answered, and treats the call accordingly.

NOTE:
Once a phone has been associated with an extension, anyone using the terminal has the capabilities of the associated station. Be sure to execute a dissociate request if the terminal can be accessed by unauthorized users. This is particularly important if you use PSA and Digital Communications Protocol (DCP) extenders to permit remote DCP access.

Before you start

Verify that Personal Station Access is set to y on the Class of Service (COS) screen.

Verify that the extension has a COS that allows PSA.

Instructions

In our example, we specify the TTI State, the Record PSA/TTI Transactions, the class of service, and the feature access codes set up PSA.

To set up Personal Station Access:

1. Type **change system-parameters features** and press RETURN.

   The Feature-Related System Parameters screen appears.

2. Complete the following fields and press ENTER.

   a. Type **voice** in the TTI State field.

   b. (Optional) Type **y** on the Record PSA/TTI Transactions in History Log field.

      These fields display only when the Terminal Translation Initialization (TTI) Enabled field on this screen is set to y.

3. Type **change cos** and press RETURN.

   The Class of Service screen appears.

4. Type **y** in the Personal Station Access (PSA) 1 field and press ENTER.

5. Type **change feature-access-codes** and press RETURN.

   The Feature Access Code (FAC) screen appears.

6. Complete the following fields and press ENTER.

   a. Type **#4** in the Personal Station Access (PSA) Associate Code field.

      This is the feature access code you will use to activate Personal Station Access at a phone.

   b. Type **#3** in the Dissociate Code field.

      This is the feature access code you will use to deactivate Personal Station Access at a phone.

More information

You can allow users to place emergency and other calls from phones that have been dissociated. To enable this, you must first assign a class of restriction (COR) for PSA-dissociated phones. You do this on the ''Feature-Related System Parameters'' screen. In addition, you must set the restrictions for this COR on the ''Class of Restriction'' screen.

If you want users to be able to place emergency calls from dissociated phones, it is also a good idea to have the system send calling party number (CPN) or automatic number identification (ANI) information for these calls. To do this, you must set the CPN, ANI for PSA Dissociated Sets field to y on the ''Feature-Related System Parameters'' screen.

Related topics

Refer to ‘‘Training users’’ on page 367 for information on how to associate or disassociate PSA.

Creating a station security code

Station Security Codes (SSC) provide security to station users by preventing other users from accessing functions associated with the user’s station. Each station user can change their own SSC if they know the station’s current settings.

You must create a system-wide SSC change feature access code (FAC) before users can change their SSC. You must also provide users with their individual SSC. A user cannot change a blank SSC.

Instructions

In our example, we set the station security code for a user.

To create a station security code:

1. Type **change feature-access-codes** and press RETURN.

   The Feature Access Code screen appears.

2. Type **#5** in the Station Security Code Change Access Code field and press ENTER.

   This sets the access codes for this features. The Command prompt appears.

3. Type **change system-parameters security** and press RETURN.

   The Security-Related System Parameters screen appears.

4. Type 4 in the Minimum Station Security Code Length field and press ENTER.

   This determines the minimum required length of the Station Security Codes you enter on the Station screen. Longer codes are more secure. If station security codes are used for external access to telecommuting features, the minimum length should be 7 or 8.

5. Type change station **1234** and press RETURN.

   This is the station extension you configured for telecommuting. The Station screen appears.

```
                                                                Page 1 of X
                                              STATION

Extension: 1234 Lock Messages? n       BCC: 0
     Type: 406D                                Security Code: 4321__    TN: 1
     Port: _____                             Coverage Path 1: ___      COR: 1
     Name: __________________________        Coverage Path 2: ___      COS: 1
                                             Hunt-to-Station: _______

STATION OPTIONS
             Data Module? n                  Personalized Ringing Pattern: 1
          Display Module? n                              Message Lamp Ext: 1014


                                               MM Complex Data Ext: ________
```

6. Type **4321** in the Security Code field and press ENTER.

Related topics

Refer to ''Station'' on page 1066 for information about and field descriptions on the Station screen.

Refer to ''Station Security Codes'' on page 1711 for a description of the Station Security Codes feature.

Assigning coverage options

Avaya MultiVantage allows you to assign two previously administered coverage paths and/or time of day coverage tables on the Station screen. This allow telecommuters to alternate between the two coverage paths and/or time of day coverage tables administered to better control how their telephone calls are handled.

For information about creating a coverage path, refer to ''Creating coverage paths'' on page 188. For information about creating a time of day coverage table, refer to ''Assigning a coverage path to users'' on page 190.

Instructions

In our example, we assign two coverage options so a user can choose from either option to control how their calls are handled.

To assign 2 coverage options:

1. Type **change feature-access-codes** and press RETURN.

   The Feature Access Code screen appears.

2. Type **#9** in the Change Coverage Access Code field and press ENTER.
3. Type **change cor 1** and press RETURN.

   The Class of Restriction screen appears.

4. In the Can Change Coverage field, type **y** and press ENTER to save your work.

   The Command prompt appears.

5. Type change station **1234** and press RETURN.

   This is the station extension you configured for telecommuting. The Station screen appears.

6. Complete the following fields:
   a. Type **2** in the Coverage Path 1 field.
   b. Type **t8** in the Coverage Path 2 field.

Related topics

Refer to ''Coverage Path'' on page 672 for information about and field descriptions on the Coverage Path screen.

Refer to ''Call Coverage'' on page 1416 for a description of the Call Coverage feature.

Refer to ''Training users'' on page 367 for information on how to alternate your coverage path option.

Refer to ''Extended User Administration of Redirected Calls'' on page 1554 for information about the Extended User Administration of Redirected Calls feature.

Setting up call forwarding

Avaya MultiVantage allows you to change your call forwarding from any on-site or off-site location.

Instructions

In our example, we assign the feature access codes and class of service to set up call forwarding. This allows your users to forward their calls to another extension.

To set up call forwarding:

1. Type **change feature-access-codes** and press RETURN.

   The Feature Access Code screen appears.

2. Set a 2-digit access code for the following fields and press ENTER.

   a. Type ***8** in the Extended Call Fwd Activate Busy D/A field.

   b. Type ***7** in the Extended Call Fwd Activate All field.

   c. Type ***6** in the Extended Call Fwd Activate Deactivation field.

   This sets the access codes for these features. The Command prompt appears.

3. Type **change cos** and press RETURN.

   The Class of Service screen appears.

4. Set the following fields to **y**.

   - Extended Forwarding All
   - Extended Forwarding B/DA

   This allows you to change the forwarding of all your calls from an off-site location.

5. Set the Restrict Call Fwd-Off Net field to **n** and press ENTER.

   This allows your users to forward calls off-site.

Interactions

- Bridged Appearance

  When the pound key (#) is pressed from a bridged appearance immediately following any of this feature’s four FACs, the system assumes that the currently active bridged extension will be administered. The station security code of the currently active bridged extension must be entered after the initial # to successfully complete the command sequence.

  If the station has only bridged appearances, the station’s extension must be dialed after the FAC to successfully complete the command sequence, since the station’s extension is not associated with any appearances.

- Distributed Communications System

  Assign a different telecommuting access extension for each switch. You can use Extended User Administration of Redirected Calls from any of the DCS nodes, but you must dial the extension of the node on which your station is defined before dialing the FAC.

- Tenant Partitioning

  The telecommuting access extension is always automatically assigned to Tenant Partition 1, so it can be accessed by all tenants.

  The tenant number of the extension being administered must be accessible by the tenant number from which the Extended User Administration of Redirected Calls FAC is dialed or the request is denied. If the FAC is dialed on site, the tenant number of the station or attendant must have access to the tenant number of the extension administered. If the FAC is dialed off site, the tenant number of the incoming trunk must have access to the tenant number of the extension administered.

Related topics

Refer to ‘‘Training users’’ on page 367 for information on how to change call forwarding.

Refer to ‘‘Call Forwarding’’ on page 1496 for a description of the Call Forwarding feature.

Assigning an extender password

Avaya MultiVantage allows you assign an extender password to a user. You can assign one password for each port on your MultiVantage solution.

Before you start

Use the Remote Extender PC in the switch room to perform this procedure.

Instructions

In our example, we set a system generated random password for John Doe.

To assign an extender password:

1. Double-click the Security icon.

   The Password Manager screen appears.

2. Double-click User Password for User 01.

3. Select Enable Password to enable the password.
4. Click random.

   This means that the password is a system generated random number. The system displays a 10-digit number in the Password field. Take note of this number, your user will need it to access the switch from home.
5. Type Doe, John and click OK.

   This is the last name and first name of the user. The system returns you to the Password Manager screen.
6. Select CommLink:Select Cards.

   A screen containing a list of cards (for example, Card A, Card B, and so on) appears. Each card corresponds to a port on your switch.
7. Select Card A and click OK.

   The system returns you to the Password Manager screen.
8. Select CommLink:Upload Password.

   The error message screen appears with the message “Administrator password not loaded”.
9. Click OK.

   The Administrator screen appears.
10. Type 123456 and click OK.

    This is the administrator’s password.
11. Select CommLink:Upload Password.

    The password is uploaded.
12. When upload is complete, click OK.

    The system returns you to the Password Manager screen.
13. Select File:Save As.

    The Save As screen appears.
14. Type doe.fil in the File field and click OK.

    The system saves the User01 information.

Installing home equipment

Avaya MultiVantage allows you to install equipment in your home so that you can utilize system facilities from off-site.

Before you start

You need the following equipment:

- MultiVantage extender remote module
- DCP sets (office and home must match)

Configure a feature access code for associating your home number to your office number. For information about configuring an associate feature access code, refer to ''Setting up Personal Station Access'' on page 354.

Instructions

Installing home equipment

To install your home equipment:

1. Plug the phone cord into the slot labeled line on the back of the module and into the wall jack.
2. Plug the phone cord into the slot labeled port on the back of the module and into the slot labeled line on the phone.
3. Plug the power cord into slot labeled power on the back of the module and the wall socket.

   The phone display Go Online appears.
4. Press 3 (Nxt).

   The phone display Set Phone Number appears.
5. Press 2 (OK) to set the phone number.
6. Type **5551234** and press DROP.

   This is the assigned analog phone number. In some areas, you may need to include your area code (for example, 3035551234). The phone display Set Phone Number appears.
7. Press 1(Prv).

   This returns you to the Go Online phone display.

8. Press 2 (OK).

   The module dials the number. When the modules connect, the phone display Enter Password appears.

9. Type **0123456789** and press DROP.

Associating your office phone number to the home station

To associate your phone number:

1. On your home station, type **#4**.

   This is the associate feature access code.

2. Type **4321** and press **#.**

   This is your extension number.

3. Type **1996** and press **#**.

   This is your password.

Disassociating your home station

To disassociate your home station:

1. Press HOLD four times.

Related topics

Refer to ‘‘Configuring Avaya MultiVantage for telecommuting’’ on page 351 for step-by-step instructions on how to configure your office equipment.

Refer to ‘‘Training users’’ on page 367 for step-by-step instructions on how to use your home station.

Setting up remote access

Remote Access permits a caller located outside the system to access the switch through the public or private network and then use the features and services of the system.

Remote Access users can dial into the system using central office (CO), Foreign Exchange (FX), Wide Area Telecommunications trunks (WATS), and Integrated Services Digital Network Primary Rate Interface (ISDN-PRI) trunks. In addition, a dedicated Remote Access Direct Inward Dialing number can be provided.

SECURITY ALERT:

Avaya has designed the Remote Access feature incorporated in this product that, when properly administered by the customer, will enable the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of remote access features. In such an event, applicable tariffs require that the customer pay all network charges for traffic. Avaya cannot be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

If you do not intend to use Remote Access now or in the future, you can permanently disable the feature. If you do decide to permanently disable the feature, it will require Avaya Services intervention to activate the feature again.

Before you start

Configure the Incoming Destination and Night Service fields on the CO trunk screen. For information about configuring a CO trunk, refer to ''Adding a CO, FX, or WATS trunk group'' on page 406.

Verify that the Authorization Codes field on the System Parameters Customer Options screen is set to **y**.

Verify that the SVN Authorization Code Violation Notification Enabled field on the Security-Related System Parameters screen is set to **y**.

Instructions

Setting up remote access

In our example, we set up a remote access extension with maximum security. This assists you in blocking unauthorized people from gaining access to your network.

To set up remote access:

1. Type **change remote-access** and press RETURN.

   The QSIG to DCS TSC Gateway screen screen appears.

```
                                  REMOTE ACCESS
   Remote Access Extension 1234___          Barrier Code Length 7__
Authorization Code Required? y                   Remote Access Dial Tone: y
Barrier Code   COR  TN  COS    Expiration Date   No. of Calls     Calls Used
1:1234567      1__  1_  1__    01/01/99          ______           ______
2:________     1__  1_  1__    __/__/__          ______           ______
3:________     1__  1_  1__    __/__/__          ______           ______
4:________     1__  1_  1__    __/__/__          ______           ______
5:________     1__  1_  1__    __/__/__          ______           ______
6:________     1__  1_  1__    __/__/__          ______           ______
7:________     1__  1_  1__    __/__/__          ______           ______
8:________     1__  1_  1__    __/__/__          ______           ______
9:________     1__  1_  1__    __/__/__          ______           ______
10:_____       1__  1_  1__    __/__/__          ______           ______
Permanently Disable? __ Disable Following A Security Violation? y
(NOTE:  You must logoff to effect permanent disabling of Remote Access)
```

2. Type **1234** in the Remote Access Extension field.

   This is the extension specified in the Incoming Destination field on the CO trunk screen.

3. Type **7** in the Barrier Code Length field.

   This is the number of digits your barrier code must be when entered.

4. Type **y** in the Authorization Code Required field.

   This means you must also enter an authorization code when you access the system's Remote Access facilities. For information about setting up access codes, refer to ''Setting up authorization codes'' on page 398.

5. Type **y** in the Remote Access Dial Tone field.

   This means you hear dial tone as a prompt to enter your authorization code.

6. Type **1234567** in the Barrier Code field.

   This is the 7-digit barrier code you must enter to access the system's Remote Access facilities.

7. Type **1** in the COR field.

   This is the class of restriction (COR) number associated with the barrier code that defines the call restriction features.

8. Type **1** in the TN field.

   This is the Tenant Partition (TN) number.

9. Type **1** in the COS field.

   This is the class of service (COS) number associated with the barrier code that defines access permissions for Call Processing features.

10. Type **01/01/99** in the Expiration Date field.

    This is the date the barrier code expires. A warning message is displayed on the system copyright screen seven days before the expiration date. The system administrator can modify the expiration date to extend the time interval, if necessary.

11. Type **y** in the Disable Following A Security Violation field.

    This disables the remote access feature following detection of a remote access security violation.

12. Press ENTER to save your work.

Disabling remote access permanently

To disable remote access permanently:

1. Type **change remote-access** and press RETURN.

   The QSIG to DCS TSC Gateway screen screen appears.

2. Type **y** in the Permanently Disable field.

   If you permanently disable this feature, it requires Avaya Services intervention to reactivate the feature. There is a charge for reactivation of this feature.

3. Press ENTER to save your work.

CAUTION:

Your attempt to disable the Remote Access feature will be lost if the switch is rebooted without saving translations. Therefore, execute a ***save translation*** *command after permanently disabling the Remote Access feature.*

More information

Remote Access provides you with access to the system and its features from the public network. This allows you to make business calls from home or use Recorded Telephone Dictation Access to dictate a letter. If authorized, you can also access system features from any on-site extension.

With Remote Access you can dial into the system using Direct Inward Dialing (DID), Central Office (CO), Foreign Exchange (FX), or 800 Service trunks. When a call comes in on a trunk group dedicated to Remote Access, the system routes the call to the Remote Access extension you have assigned. If DID is provided and the Remote Access extension is within the range of numbers that can be accessed by DID, Remote Access is accessed through DID.

Barrier codes provide your system security and define calling privileges through the administered COR. You can administer up to 10 barrier codes, each with a different COR and COS. Barrier codes can be from 4 to 7 digits, *but all codes must be the same length*. You can also require that users enter an authorization code to use this feature. Both barrier codes and authorization codes are described under ‘‘Setting up authorization codes’’ on page 398.

Related topics

Refer to ‘‘QSIG to DCS TSC Gateway screen’’ on page 1029 for information about and field descriptions on the Remote Access screen.

Refer to ‘‘Remote Access’’ on page 1680 for a description of the Remote Access feature.

Training users

DEFINITIY ECS allows you to associate and disassociate PSA, change the coverage path for your station, change the extension to which you forward your calls, and change your personal station’s security code.

Before you start

Configure PSA. For information about configuring PSA, refer to ‘‘Setting up Personal Station Access’’ on page 354

Assign two coverage options for your system. For information on how to assign coverage options, refer to ‘‘Assigning coverage options’’ on page 357.

Configure call forwarding for your system. For information about configuring call forwarding, refer to ‘‘Setting up call forwarding’’ on page 358.

Configure security codes for a station. For information about configuring personal station security codes, refer to ‘‘Assigning an extender password’’ on page 360.

Instructions

Associating PSA

In this example, we associate PSA (preferences and permissions) assigned to your station with another compatible terminal.

To associate PSA:

1. Dial **#4**.

   This is the associate PSA feature access code. You hear dial tone.

2. Type **1234** and press #.

   This is your extension.

3. Type **4321** and press #.

   This is your Station Security Code. You hear a confirmation tone.

Disassociating PSA

In our example, we disassociate PSA from the station you are using.

To disassociate PSA:

1. Dial **#3**.

   This is the disassociate PSA feature access code. You are no longer PSA associated to this station.

Changing a coverage option

In this example, we change the coverage option from path 1 to path 2 from a remote location.

To change a coverage option:

1. Dial **1234.**

   This is the extension you configured for telecommuting. You hear dial tone.

2. Dial **#9** and press #.

   This is the feature access code you set for changing a coverage path. You hear dial tone.

3. Dial **4321** and press #.

   This is the extension for which you want to change the coverage path.

4. Dial **87654321** and press #.

   This is the extension security code.

5. Dial **2**.

   This is the new coverage path. You hear confirmation tone.

Changing call forwarding

In this example, we change call forwarding to extension 1235.

To change call forwarding:

1. Dial **1234.**

   This is the extension you configured for telecommuting.

2. Dial **#8** and press #.

   This is the feature access code you set for activating extended call forward. You hear dial tone.

3. Dial **4321**and press #.

   This is the extension from which you want to forward calls.

4. Dial **87654321** and press #.

   This is the extension security code. You hear dial tone.

5. Dial **1235**.

   This is the extension to which you want to forward calls. You hear the confirmation tone.

Changing your personal station security codes

In this example, we change the security code for extension 1235 from 98765432 to 12345678.

To change your security code:

1. Dial **#5**.

   This is the feature access code you set for changing your security code. You hear dial tone.

2. Dial **1235** and press #.

   This is the extension for which you want to change the security code.

3. Dial **98765432** and press #.

   This is the current security code for the extension. You hear dial tone.

4. Dial **12345678** and press #.

   This is the new security code. Security codes can be 4- to 8-digits long.

5. Dial **12345678** and press #.

   This is to confirm your new security code. You hear the confirmation tone.

   NOTE:
   If you cannot change your security code, Manager 1 can clear the problem using the Clear Audit Summary command.

Interrupting the command sequence for personal station security codes

To interrupt the command sequence for personal station security codes:

1. To interrupt the command sequence before step 3, choose one of these options:
   - Hang up or press the disconnect or recall button before hearing intercept tone in step 3.

     The system does not log an invalid attempt. You must restart the process at step 1.
   - Type ***** before the second # in step 3.

     You must begin the change sequence at the point of entering your extension in step 2. (You should not enter the FAC again.)
   - Type ***** after the FAC has been entered and before the final #.

     You must restart the process at step1.
2. To interrupt the command sequence after step 3, type ***** in steps 4 or 5, you must begin the change sequence at the point of entering the new SSC in step 4.

   If you hear intercept tone in any step, the command sequence has been invalidated for some reason and you must restart the process at step 1.

   If you hear intercept tone after step 3, the system logs an invalid attempt via the Security Violation Notification (SVN) feature. This is true even if you attempt to interrupt the change sequence with an asterisk.

Enhancing system security

11

Toll fraud is the theft of long distance service and can come from both internal and external sources. When toll fraud occurs, your company is responsible for usage charges. In addition, unauthorized use may tie up your system, preventing your customers from reaching you and your employees from doing business.

Avaya designed Avaya MultiVantageMultiVantage to help you to limit toll fraud. However, there are steps that you, as the administrator, must also take to keep your system secure from unauthorized use.

Need help quickly?

- For assistance with toll fraud prevention (including systems and products), call the Avaya Toll Fraud Intervention Hotline at 800-643-2353 or contact your Avaya representative.
- If you have identified fraudulent calling in progress, and require assistance in stopping the fraud, call the Avaya Technical Service Center at 800-242-2121 and select the toll fraud help option or contact your Avaya representative.

Basic security

Keeping your system secure

The following is a partial list you can use to help secure your system. It is not intended as a comprehensive security checklist. Refer to the *Avaya Products Security Handbook* for more information about these and other security-related features.

- Secure the system administration and maintenance ports and/or logins on Avaya MultiVantage using the Access Security Gateway. This optional password authentication interface program is provided to customers with maintenance contracts.
- Activate Security Violation Notification to report unsuccessful attempts to access the system. Security Violation Notification lets you automatically disable a valid login ID following a security violation involving that login ID and disable remote access following a security violation involving a barrier code or authorization code.
- Use the **list history** command to determine if unauthorized changes have been made to the system. To assist in identifying unauthorized use of the system, the History report lists each time a user logs on or off the system. Refer to the *Reports for Avaya MultiVantage™ Software* for more information about this report.
- Secure trunks using Automatic Route Selection, Class of Restriction, Facility Restriction Levels and Alternate Facility Restriction Levels, Authorization Codes, Automatic Circuit Assurance, and Forced Entry of Account Codes (refer to ''Call Detail Recording'' on page 1437 for more information).
- Activate Enhanced Call Transfer for your voice messaging system, if available. This limits transfers to valid extensions, but you also need to restrict transfers to extensions that may offer dial tone to the caller, such as screen extensions.

Preventing toll fraud

Top 15 tips to help prevent toll fraud

1. Protect system administration access

   Make sure secure passwords exist for all logins that allow System Administration or Maintenance access to the system. Change the passwords frequently.

   Set logoff notification and forced password aging when administering logins. You must assign passwords for these logins at setup time.

   Establish well-controlled procedures for resetting passwords.

2. Prevent voice mail system transfer to dial tone

   Activate “secure transfer” features in voice mail systems.

   Place appropriate restrictions on voice mail access/egress ports.

   Limit the number of invalid attempts to access a voice mail to five or less.

3. Deny unauthorized users direct inward system access (screen)

   If you are not using the Remote Access features, deactivate or disable them.

   If you are using Remote Access, require the use of barrier codes and/or authorization codes set for maximum length. Change the codes frequently.

   It is your responsibility to keep your own records regarding who is allowed to use which authorization code.

4. Place protection on systems that prompt callers to input digits

   Prevent callers from dialing unintended digit combinations at prompts.

   Restrict auto attendants and call vectors from allowing access to dial tone.

5. Use system software to intelligently control call routing

   Create Automatic Route Selection or World Class Routing patterns to control how each call is to be handled.

   Use “Time of Day” routing capabilities to limit facilities available on nights and weekends.

   Deny all end-points the ability to directly access outgoing trunks.

6. Block access to international calling capability

   When international access is required, establish permission groups.

   Limit access to only the specific destinations required for business.

7. Protect access to information stored as voice

   Password restrict access to voice mail mailboxes.

   Use non-trivial passwords and change passwords regularly.

8. Provide physical security for telecommunications assets

   Restrict unauthorized access to equipment rooms and wire connection closets.

   Protect system documentation and reports data from being compromised.

9. Monitor traffic and system activity for abnormal patterns

   Activate features that “turn off” access in response to unauthorized access attempts.

   Use Traffic and Call Detail reports to monitor call activity levels.

10. Educate system users to recognize toll fraud activity and react appropriately

    From safely using calling cards to securing voice mailbox password, train your users on how to protect themselves from inadvertent compromises to the system’s security.

11. Monitor access to the dial-up maintenance port. Change the access password regularly and issue it only to authorized personnel. Consider activating Access Security Gateway.

12. Create a switch system management policy concerning employee turnover and include these actions:

    a. Delete any unused voice mailboxes in the voice mail system.

    b. Immediately delete any voice mailboxes belonging to a terminated employee.

    c. Immediately remove the authorization code if a terminated employee had screen calling privileges and a personal authorization code.

    d. Immediately change barrier codes and/or authorization codes shared by a terminated employee. Notify the remaining users of the change.

    e. Remove a terminated employee’s login ID if they had access to the system administration interface. Change any associated passwords immediately.

13. Back up system files regularly to ensure a timely recovery. Schedule regular, off-site backups.

14. Callers misrepresenting themselves as the “phone company,” “AT&T,” “RBOCS,” or even known employees within your company may claim to be testing the lines and ask to be transferred to “900,” “90,” or ask the attendant to do “start 9 release.” This transfer reaches an outside operator, allowing the unauthorized caller to place a long distance or international call. Instruct your users to never transfer these calls. Do not assume that if “trunk to trunk transfer” is blocked this cannot happen.
15. Hackers run random generator PC programs to detect dial tone. Then they revisit those lines to break barrier codes and/or authorization codes to make fraudulent calls or resell their services. They do this using your telephone lines to incur the cost of the call. Frequently these call/sell operations are conducted at public payphones located in subways, shopping malls, or airport locations. Refer to ‘‘QSIG to DCS TSC Gateway screen’’ on page 1029 to prevent this happening to your company.

Physical security

Physical security is your responsibility. Implement the following safeguards as an added layer of security:

1. Unplug and secure attendant console handsets when the attendant position is not in use.
2. Lock wiring closets and switch rooms.
3. Keep a log book register of technicians and visitors.
4. Shred all switch information or directories you discard.
5. Always demand verification of a technician or visitor by asking for a valid I.D. badge.
6. Keep any reports that may reveal trunk access codes, screen barrier codes, authorization codes, or password information secure.
7. Keep the attendant console and supporting documentation in an office that is secured with a changeable combination lock. Provide the combination only to those individuals who need to enter the office.
8. Keep any documentation pertaining to switch operation secure.
9. Label all backup tapes or flash cards with correct dates to avoid using an outdated one when restoring data. Be sure that all backup media have the correct generic software load.

System security checklist

Here's some of the steps required for indemnification. Use these to analyze your system security.

1. Remove all default factory logins of **cust**, **rcust**, **browse**, **nms**, and **bcms** and assign unique logins with 7-character alphanumeric passwords and a 90-day password aging. Use the **list logins** command to find out what logins are there.

2. If you do not use Remote Access, be sure to disable it permanently.

   Tip:
   *You can use the **display remote-access** command to check the status of your remote access.*

   To disable Remote Access, on the Remote Access screen, Permanently Disable field, type **y**. Refer to ''QSIG to DCS TSC Gateway screen'' on page 1029 for more information on remote access.

   NOTE:
   Avaya recommends that you permanently disable Remote Access using the **change remote-access** command. If you do permanently disable Remote Access, the code is removed from the software. Avaya charges a fee to restore the Remote Access feature.

3. If you use Remote Access, but only for internal calls, change announcements or remote service observing.

   a. Use a 7-digit barrier code.

   b. Assign a unique Class of Restriction (COR) to the 7-digit barrier code.

      The unique COR must be administered where the FRL is **0**, the Calling Party Restriction field is **outward**, the Calling Permissions field is **n** on all unique Trunk Group COR.

   c. Assign Security Violation Notification Remote to **10** attempts in **2** minutes.

   d. Set the aging cycle to **90** days with **100** call limit per barrier code.

   Refer to ''QSIG to DCS TSC Gateway screen'' on page 1029 for more information.

4. If you use Remote Access to process calls off-net or in any way access the public network:

    a. Use a 7-digit barrier code.

    b. Assign a unique COR to the barrier code.

    c. Restrict the COR assigned to each barrier code by FRL level to only the required calling areas to conduct business.

    d. Set the aging cycle to **90** days with **100** call limit per barrier code.

    e. Suppress dial tone where applicable.

    f. Administer Authorization Codes.

    g. Use a minimum of 11 digits (combination of barrier codes and authorization codes).

    h. Assign Security Violation Notification Remote to 10 attempts in 2 minutes.

5. If you use vectors:

    a. Assign all Vector Directory Numbers (VDN) a unique COR. Refer to *Avaya MultiVantage™ Call Center Software Release 11 Guide to ACD Call Centers* for more information.

NOTE:
The COR associated with the VDN dictates the calling privileges of the VDN/vector. High susceptibility to toll fraud exists on vectors that have “collect digits” steps. When a vector collects digits, it processes those digits back to the switch and if the COR of the VDN allows it to complete the call off-net, it will do so. For example, the announcement “If you know your party’s 4-digit extension number, enter it now” results in 4 digits being collected in step 6. If you input “90##” or “900#”, the 4 digits are analyzed and if “9” points towards ARS and “0” or “00” is assigned in the ARS Analysis Tables and the VDN COR allows it, the call routes out of the switch to an outside local exchange or long distance operator. The operator then connects the call to the requested number.

    b. If vectors associated with the VDN do not require routing the call off-net or via AAR, assign a unique COR where the FRL is **0**, the Calling Party Restriction field is **outward**, the Calling Permissions field is **n** on all unique Trunk Group COR.

c. If the vector has a “route-to” step that routes the call to a remote switch via AAR, assign a unique COR with a unique ARS/AAR Partition Group, the lowest FRL to complete an AAR call, and **n** on all unique COR assigned to your public network trunking facilities on the Calling Permissions. Assign the appropriate AAR route patterns on the AAR Partition Group using the **change aar analysis partition x 2** command.

Tip:
You can use the ***display aar analysis print*** *command to print a copy of your Automatic Alternate Routing (AAR) setup before making any changes. You can use the printout to correct any mistakes.*

d. If the vector has a “route-to” step that routes the call to off-net, assign a unique COR with a unique ARS/AAR Partition Group, the lowest FRL to complete an ARS call, and **n** on all unique COR assigned to your public network trunking facilities on the Calling Permissions. Assign the appropriate complete dial string in the “route-to” step of the vector the unique ARS Partition Group using the **change ars analysis partition x 2** command.

6. On the Feature Access Code screen, Facility Test Calls Access Code, the Data Origination Access Code, and the Data Privacy Access Code fields, change from the default or remove them.

NOTE:
These codes, when dialed, return system dial tone or direct access to outgoing trunking facilities. Transfers to these codes can take place via an unsecured vector with “collect digits” steps or an unsecured voice mail system.

7. Restrict Call Forwarding Off Net on every class of service.

Refer to ‘‘Class of Service’’ on page 650 for more information on Class of Service.

NOTE:
You cannot administer loop-start trunks if Call Forwarding Off Net is required.

8. If loop start trunks are administered in the switch and cannot be changed by the Local Exchange Company, block all class of service from forwarding calls off-net. In the Class of Service screen, Restriction Call Fwd-Off Net field, set to **y** for the 16 (0-15) COS numbers.

   Refer to ''Class of Service'' on page 650 for more information.

   NOTE:
   If a station is call forwarded off-net and an incoming call to the extension establishes using a loop-start trunk, incorrect disconnect supervision can occur at the Local Exchange Central Office when the call terminates. This gives the caller recall or transfer dial tone to establish a fraudulent call.

9. Administer Call Detail Recording on all trunk groups to record both incoming and outgoing calls.

   Refer to ''Collecting information about calls'' on page 525 for more information.

10. On the ''Route Pattern'' on page 1039, be careful assigning route patterns with an FRL of **0**; these allow access to outgoing trunking facilities. Avaya recommends assigning routes with an FRL of 1 or higher.

    NOTE:
    An exception might be assigning a route pattern with an FRL of 0 to be used for 911 calls so even restricted users may dial this in emergencies.

    Tip:
    You can use the ***list route-pattern print*** *command to print a copy of your facility restriction levels (FRL) and check their status.*

11. On all trunk group screens, set the Dial Access field to **n**. If set to **y**, it allows users to dial Trunk Access Codes, thus bypassing all the ARS call screening functions.

    Refer to ''Trunk Group'' on page 1167 for more information.

12. On the ''AAR and ARS Digit Analysis Table'' on page 559, set all dial strings not required to conduct business to **den** (deny).

13. If you require international calling, on the ''AAR and ARS Digit Conversion Table'' on page 564, use only the 011+ country codes/city codes or specific dial strings.

14. Assign all trunk groups or same trunk group types a unique Class of Restriction. If the trunk group does not require networking through your switch, administer the Class of Restriction of the trunk group where the FRL is **0**, the Calling Party Restriction field is **outward**, and all unique Class of Restriction assigned to your outgoing trunk groups are **n**. Refer to ''Class of Restriction'' on page 636 for more information.

   Tip:
   *You can use the **list trunk-group print** command to have a printout of all your trunks groups. Then, you can use the **display trunk-group x** command (where x is the trunk group) to check the Class of Restriction (COR) of each trunk group.*

15. For your AUDIX, on the System Appearance screen, set:

   - the Enhanced Call Transfer field to **y**.
   - the Transfer Type field to **enhanced**. If set to **basic**, set the Transfer Restriction field to **subscribers**. Refer to ''Feature-Related System Parameters'' on page 766 for more information.

   NOTE:
   The Class of Restriction of the voice mail ports dictates the calling restrictions of the voice mail. If the above settings are not administered correctly, the possibility exists to complete a transfer to trunk access codes or ARS/AAR feature codes for fraudulent purposes. Never assign mailboxes that begin with the digits or trunk access codes of ARS/AAR feature access codes. Require your users to use a mailbox password length greater than the amount of digits in the extension number.

16. Avaya recommends you administer the following on all voice mail ports:

   - Assign all voice mail ports a unique Class of Restriction. Refer to ''Class of Restriction'' on page 636 for more information.
   - If you are not using outcalling, fax attendant, or networking, administer the unique Class of Restriction where the FRL is **0**, the Calling Party Restriction field is **outward**, and all unique trunk group Class of Restriction on the Calling Permissions are **n**. Refer to ''Class of Restriction'' on page 636 for more information.

   NOTE:
   Avaya recommends you administer as many layers of security as possible. You can implement steps 9 and 16 as a double layer of security. In the event that the voice mail becomes unsecured for any reason, the layer of security on the switch takes over, and vice versa.

17. Administer all fax machines, modems, and answering machines analog voice ports as follows:
    - Set the Switchhook Flash field to **n**.
    - Set the Distinctive Audible Alert field to **n**. Refer to ''Station'' on page 1066 for more information.
18. Install a Call Accounting System to maintain call records. In the CDR System Parameters screen, Record Outgoing Calls Only field, set to **y**. Refer to ''CDR System Parameters'' on page 624 for more information.

    NOTE:
    Call Accounting Systems produce reports of call records. It detects phones that are being hacked by recording the extension number, date and time of the call, and what digits were dialed.

Adding logins and passwords

This section shows you how to add a user and their password. To add a login, you must be a superuser with authority to administer permissions.

When adding logins, remember the following:

- Type the new login name as part of the **add** command. The name must be 3–6 alphanumeric characters in length, and can contain the characters 0-9, a-z, A-Z.
- The password must be from 7 to 11 alphanumeric characters in length and contain at least 1 non-alphabetic character.

Instructions

We will add the login **angi3** with the password **b3stm0m**. We also will require the user to change their password every 30 days.

To add new logins and passwords:

1. Type **add login angi3** and press RETURN.

   The Login Administration screen appears.

```
                              LOGIN ADMINISTRATION

                 Password of Login Making Change:

             LOGIN BEING ADMINISTERED
                                Login’s Name: angi3
                                  Login Type:
                               Service Level:
    Disable Following a Security Violation?
                                                          Access to INADS Port? _

            LOGIN’S PASSWORD INFORMATION
                            Login’s Password:
                    Reenter Login’s Password:
        Password Aging Cycle Length (Days): 30

        LOGOFF NOTIFICATION
          Facility Test Call Notification? y     Acknowledgment Required? y
               Remote Access Notification? y     Acknowledgment Required? y

ACCESS SECURITY GATEWAY PARAMETERS
Access Security Gateway? n
```

The Login’s Name field shows the name you typed in the **add** command.

2. In the Password of Login Making Change field, type your superuser password.
3. In the Disable Following a Security Violation field, type **y** to disable this login following a login security violation.

   This field appears only if on the Security-Related System Parameters screen, SVN Login Violation Notification field is **y**.
4. In the Login’s Password field, type **b3stm0m**.

   The password does not appear on the screen as you type.
5. In the Reenter Login’s Password field, retype **b3stm0m**.
6. In the Password Aging Cycle Length (Days) field, type **30**.

   This requires the user to change the password every 30 days.
7. Press ENTER to save your changes.

   Now you need to set the permissions for this new login.

8. Type **change permissions angi3** and press RETURN.

   The Command Permission Categories screen appears.

```
                        COMMAND PERMISSION CATEGORIES
                          Login Name: angi3
COMMON COMMANDS
               Display Admin. and Maint. Data? n
                          System Measurements? n

ADMINISTRATION COMMANDS
      Administer Stations? y              Administer Features? n
        Administer Trunks? n           Administer Permissions? n
  Additional Restrictions? y

MAINTENANCE COMMANDS
        Maintain Stations? n        Maintain Switch Circuit Packs? n
          Maintain Trunks? n       Maintain Process Circuit Packs? n
         Maintain Systems? n                Maintain Enhanced DS1? n
```

9. In the Administer Stations field, type **y**.

   This allows your user to add, change, duplicate, or remove stations, data modules and associated features.

10. In the Additional Restrictions field, type **y**.

    A **y** in this field brings up the second and third pages of this screen.

```
                        COMMAND PERMISSION CATEGORIES
                           RESTRICTED OBJECT LIST
        vdn                             ______________________
        _______________________         ______________________
        _______________________         ______________________
        _______________________         ______________________
        _______________________         ______________________
        _______________________         ______________________
        _______________________         ______________________
        _______________________         ______________________
        _______________________         ______________________
        _______________________         ______________________
```

11. In the first field, type **vdn**.

    This restricts your user from administering a VDN.

12. Press ENTER to save your changes.

More information

When you add a login, the Security Measurement reports do not update until the next hour.

Password aging is an option you can start while administering logins. The password for each login can be aged starting with the date the password was created or changed and continuing for a specified number of days (1 to 99).

The system notifies the user at the login prompt, 7 days before the password expiration date, their password is about to expire. When the password expires, the user needs to enter a new password into the system before logging in.

Changing a login

This section shows you how to change a user's login. You may need to change a user's password because it has expired. To change a login's attributes, you must be a superuser with authority to administer permissions.

When changing logins, remember the following:

- Type the new login name as part of the change command. The name must be 3–6 alphanumeric characters in length, and can contain the characters 0-9, a-z, A-Z.
- The password must be from 7 to 11 alphanumeric characters in length and contain at least 1 non-alphabetic character.

Instructions

We will change the login **angi3** with the password **b3stm0m**. We also will require the user to change their password every 30 days.

To change logins:

We will change the login **angi3**.

1. Type **change login angi3** and press RETURN.

The Login Administration screen appears.

```
                              LOGIN ADMINISTRATION

                Password of Login Making Change:

            LOGIN BEING ADMINISTERED
                               Login's Name:angi3
                                 Login Type:
                            Service Level:
    Disable Following a Security Violation?
                                                     Access to INADS Port? _
LOGIN'S PASSWORD INFORMATION
                          Login's Password:
                  Reenter Login's Password:
        Password Aging Cycle Length (Days):

        LOGOFF NOTIFICATION
          Facility Test Call Notification? y     Acknowledgment Required? y
               Remote Access Notification? y     Acknowledgment Required? y

ACCESS SECURITY GATEWAY PARAMETERS
Access Security Gateway? n
```

2. In the Password of Login Making Change field, type your superuser password.
3. In the Login's Password field, type **b3stm0m**.

   This is the login for the password you are changing.
4. In the Reenter Login's Password field, retype **b3stm0m**.

   The password does not appear on the screen as you type.
5. In the Password Aging Cycle Length (Days) field, type **30**.

   This requires the user to change the password every 30 days.
6. Press ENTER to save your changes.

Related topics

"Logging into the system".

Displaying a login

This section shows you how to display a user's login and review their permissions.

Instructions

To display a login such as **angi3**:

1. Type **display login angi3** and press RETURN.

   The Login Administration appears and displays all information about the requested login except the password.

Removing a login

This section shows you how to remove a user's login. To remove a login, you must be a superuser.

Instructions

To remove a login such as **angi3**:

1. Type **remove login angi3** and press RETURN.

   The Login Administration screen appears showing information for the login you want to delete.

2. Press ENTER to remove the login, or press CANCEL to leave this screen without removing the login.

More information

When you remove a login, the Security Measurement reports do not update until the next hour.

Related topics

"Logging into the system".

Using access security gateway

This section shows you how to use Access Security Gateway (ASG). ASG prevents unauthorized access by requiring the use of the hand-held Access Security Gateway Key for logging into the system.

You need superuser privileges to perform any of the ASG procedures.

Before you start

You need an Access Security Gateway Key.

On the ''System-Parameters Customer-Options'' screen, verify the Access Security Gateway (ASG) field is **y**. If not, contact your Avaya representative.

Instructions

To set up access security gateway:

1. Type **change login xxxx** and press RETURN, where xxxx is the alphanumeric login ID.

   The Login Administration screen appears.

```
                        LOGIN ADMINISTRATION

                  Password of Login Making Change:

              LOGIN BEING ADMINISTERED
                                 Login's Name:xxxxxxx
                                   Login Type:
                                Service Level:
     Disable Following a Security Violation?
                                                          Access to INADS Port? _

             LOGIN'S PASSWORD INFORMATION
                             Login's Password:
                     Reenter Login's Password:
         Password Aging Cycle Length (Days):

         LOGOFF NOTIFICATION
           Facility Test Call Notification? y     Acknowledgment Required? y
                Remote Access Notification? y     Acknowledgment Required? y

 ACCESS SECURITY GATEWAY PARAMETERS
 Access Security Gateway? n
```

2. In the Password of Login Making Change field, type your password.
3. In the Access Security Gateway field, type **y**.

   When set to **y**, the Access Security Gateway Login Administration screen (page 2) appears automatically.
4. Either:
   - Set the System Generated Secret Key field to:
     - **y** for a system-generated secret key, or
     - **n** for a secret key to be entered by the administrator, *or*
   - In the Secret Key field, enter your secret key.

     Be sure to remember your secret key number.
5. All other fields on page 2 are optional.
6. Press ENTER to save your changes.
7. Type **change system-parameters security** and press RETURN.

   The Security-Related System Parameters screen appears.

```
                                                         Page  2  of  2
                SECURITY-RELATED SYSTEM PARAMETERS

SECURITY VIOLATION NOTIFICATION PARAMETERS

  SVN Station Security Code Violation Notification Enabled? y
          Originating Extension: ________   Referral Destination: ________
Station Security Code Threshold: 10                Time Interval: 0:03
         Announcement Extension: ________

STATION SECURITY CODE VERIFICATION PARAMETERS

                     Minimum Station Security Code Length: 4
  Security Code for Terminal Self Administration Required? y

ACCESS SECURITY GATEWAY PARAMETERS

  SYSAM-LCL? n      SYSAM-RMT? y
      MAINT? n       SYS-PORT? n
```

8. In the Access Security Gateway Parameters section, you determine which of the following necessary port type fields to set to **y**.

   NOTE:
   Avaya recommends that you protect the SYSAM-RMT port since it is a dial-up port and therefore is more susceptible to compromise.

   In our example, in the SYSAM-RMT field, we'll type **y**.
9. Press ENTER to save your changes.

Disabling Access Security Gateway

To temporarily disable ASG while users are on vacation or travel:

1. Type **change login xxxx** and press RETURN, where xxxx is the alphanumeric login ID.

   The Login Administration screen appears.

```
                      LOGIN ADMINISTRATION

                Password of Login Making Change:

            LOGIN BEING ADMINISTERED
                               Login's Name:xxxxxxx
                                 Login Type:
                              Service Level:
     Disable Following a Security Violation?
                                                          Access to INADS Port? _

           LOGIN'S PASSWORD INFORMATION
                           Login's Password:
                   Reenter Login's Password:
         Password Aging Cycle Length (Days):

         LOGOFF NOTIFICATION
           Facility Test Call Notification? y     Acknowledgment Required? y
                Remote Access Notification? y     Acknowledgment Required? y

 ACCESS SECURITY GATEWAY PARAMETERS
 Access Security Gateway? n
```

2. On the Access Security Gateway Login Administration page (page 2), set the Blocked field to **y**.

   Setting the Blocked field to **y** does not remove the login from the system, but temporarily disables the login.

3. Press ENTER to save your changes.

   NOTE:
   A superuser can disable and restart access for another superuser.

Restarting Access Security Gateway

To restart temporarily disabled access security gateway access for login:

1. Type **change login xxxx** and press RETURN, where xxxx is the alphanumeric login ID.

   The Login Administration screen appears.

2. On the Access Security Gateway Login Administration page (page 2), set the Blocked field to **n**.

3. Press ENTER to save your changes.

Loss of an ASG key

If a user loses their Access Security Gateway Key:

1. Modify any logins associated with the lost Access Security Gateway Key. Refer to the *Access Security Gateway Key User's Guide* to change your PIN.
2. If the login is no longer valid, type **remove login xxxx** and press RETURN, to remove the invalid login from the system, where xxxx is the alphanumeric login ID.
3. To keep the same login, change the Secret Key associated with the login to a new value.
4. Using the new secret key value, re-key devices that generate responses and interact with the login.

Monitoring the Access Security Gateway history log

The Access Security Gateway Session History Log records all ASG session establishment and session rejection events except when, on the Login Administration screen, the Access to INADS Port field is **y**. You must be a superuser to use the **list asg-history** command.

1. Type **list asg-history** and press RETURN.

   The Access security gateway screen appears.

```
                         ACCESS SECURITY GATEWAY SESSION HISTORY

Date     Time     Port         Login     Status
01/06    12:45    SYSAM-RMT    csand     AUTHENTICATED
01/05    01:32    SYSAM-LCL    jsmith    REJECT-BLOCK
01/05    12:33    SYSAM-RMT    ajones    REJECT-EXPIRE
01/03    15:10    SYSAM-RMT    swrigh    REJECT-PASSWORD
01/02    08:32    SYSAM-LCL    jsmith    REJECT-INVALID
01/02    07:45    SYSAM-RMT    mehrda    REJECT-RESPONSE
```

This screen contains the following fields:

- Date — Contains the date of the session establishment or rejection. For example, the date displays in the mm/dd format where mm = month and dd = day.
- Time — Contains the time of the session establishment or rejection. For example, the time displays in the hh/mm format where hh = hour and mm = minute.

- Port — Contains the port mnemonic associated with the port on which the session was established or rejected. The port mnemonics for G3r systems are SYSAM-LCL, SYSAM-RMT, MAINT, and SYS-PORT. For G3si systems, they are MRG1, INADS, NET, and EPN.
- Login — Contains the alphanumeric login string entered by the user and associated with the session establishment or rejection.
- Status — Contains a code that indicates whether the session was established or rejected and, if rejected, the reason for the rejection. Refer to Access security gateway for a list of the possible status values.

Related topics

‘‘Logging in with Access Security Gateway’’ on page 33

‘‘Security violations notification’’ on page 1693

Changing login permissions

This section shows you how to change login permissions.

Once you have created a login, you can modify the permissions associated with the login. The system maintains default permissions for each level of login, but you may want to further restrict the login, or at least make sure the defaults are appropriate for the user. The default values for these fields vary based on the login type.

Instructions

We will change the login permissions of **angi3**.

To change login permissions:

1. Type **change permissions angi3** and press RETURN.

The Command Permission Categories screen appears.

```
                COMMAND PERMISSION CATEGORIES
                           Login Name: angi3
COMMON COMMANDS
               Display Admin. and Maint. Data? n
                          System Measurements? n

ADMINISTRATION COMMANDS
      Administer Stations? y            Administer Features? n
        Administer Trunks? n         Administer Permissions? n
  Additional Restrictions? y

MAINTENANCE COMMANDS
        Maintain Stations? n      Maintain Switch Circuit Packs? n
          Maintain Trunks? n     Maintain Process Circuit Packs? n
         Maintain Systems? n              Maintain Enhanced DS1? n
```

2. In the Administer Stations field, type **y**.

   This allows your user to add, change, duplicate, or remove stations, data modules and associated features.

3. In the Additional Restrictions field, type **y**.

   A **y** in this field brings up the second and third pages of this screen.

```
                         COMMAND PERMISSION CATEGORIES
                            RESTRICTED OBJECT LIST
        vdn                            ______________________
        _______________________        ______________________
        _______________________        ______________________
        _______________________        ______________________
        _______________________        ______________________
        _______________________        ______________________
        _______________________        ______________________
        _______________________        ______________________
        _______________________        ______________________
        _______________________        ______________________
```

4. In the first field, type **vdn**.

   This restricts your user from administering a VDN.

5. Press ENTER to save your changes.

Changing passwords

This section shows you how to change a user's password.

Instructions

We will change the password for login **angi3** to **g3or5e**.

To change passwords:

1. Type **change password angi3** and press RETURN.

   The Password Administration screen appears.

```
                              PASSWORD ADMINISTRATION

Password of Login Making Change: angi3

LOGIN BEING CHANGED
                      Login Name:

LOGIN'S PASSWORD INFORMATION
                Login's Password:
        Reenter Login's Password:
```

2. In the Password of Login Making Change field, type your password to change any field on this screen.

   We'll type **angi3**.

3. In the Login's Password field, type the initial password for this login.

   We'll type **g3or5e**.

   Notify the owner of the login to change their password immediately. The password does not appear on the screen as you type.

   A password must be from 4 to 11 characters in length and contain at least 1 alphabetic and 1 numeric symbol.

4. In the Reenter Login's Password field, retype the login's password as above, for verification.

   We'll type **g3or5e**.

   The password does not appear on the screen as you type.

5. Press ENTER to save your changes.

Using busy verify

This section shows you how to use Busy Verify (also known as Busy Verification) to help find fraud problems.

When you suspect toll fraud, you can interrupt the call on a specified trunk group or extension number and monitor the call in progress. Callers will hear a long tone to indicate the call is being monitored.

SECURITY ALERT:

Listening to someone else's calls may be subject to federal, state, or local laws, rules, or regulations. It may require the consent of one or both of the parties on the call. Familiarize yourself with all applicable laws, rules, and regulations and comply with them when you use this feature.

Before you start

On the Trunk Group screen, verify the Dial Access field is **y**. If not, contact your Avaya representative.

Instructions

To use busy verify:

1. Type **change station xxxx** and press RETURN, where xxxx is the station to be assigned the busy verify button.

   For our example, we'll enter extension **1014**.

   The Station screen appears.

```
                                                                    Page  3  of  X
                                        STATION
     SITE DATA
          Room: _______                                            Headset? n
          Jack: ___                                                Speaker? n
         Cable: ___                                               Mounting: d
         Floor: _______                                       Cord Length: 0_
      Building: _______                                         Set Color: _____

ABBREVIATED DIALING
     List1: ________                List2: _________           List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                        4: verify__
 2: call-appr                        5: ________
 3: call-appr
```

2. In the Feature Button Assignments area, type **verify**.
3. Press ENTER to save your changes.
4. To activate the feature, press the VERIFY button on the phone and then enter the Trunk Access Code and member number to be monitored.

Setting up security violations notification

This section shows you how to use Security Violations Notification (SVN) to set security-related parameters and to receive notification when established limits are exceeded. You can run reports related to invalid access attempts. You also can disable a login ID or remote access authorization that is associated with a security violation.

When a security violation has occurred, there are steps that you can take to be sure that this same attempt is not successful in the future. Refer to the *Avaya Products Security Handbook* for more information.

Before you start

If you are using ASG, on the System-Parameters Customer-Options screen, verify the Access Security Gateway (ASG) field is **y**. If not, contact your Avaya representative.

Instructions

To set up security violations notification:

1. Type **change system-parameters security** and press RETURN.

The Security-Related System Parameters screen appears.

```
                 SECURITY-RELATED SYSTEM PARAMETERS

SECURITY VIOLATION NOTIFICATION PARAMETERS

  SVN Login Violation Notification Enabled? y
        Originating Extension: 3040_ Referral Destination: attd_
              Login Threshold: 3_          Time Interval: 0:03
       Announcement Extension: ________

  SVN Remote Access Violation Notification Enabled? y
        Originating Extension: 2719_ Referral Destination: 2720_
       Barrier Code Threshold: 3           Time Interval: 0:03
       Announcement Extension: ________

  SVN Authorization Code Violation Notification Enabled? y
        Originating Extension: 3030_ Referral Destination: 3031_
 Authorization Code Threshold: 3           Time Interval: 0:03
       Announcement Extension: ________
```

2. In the SVN Login Violation Notification Enabled field, type **y**.

   This sets Security Violations Notification login violation notification.

3. In the Originating Extension field, type **3040**.

   This becomes the phone extension for the purpose of originating and identifying SVN referral calls for login security violations.

4. In the Referral Destination field, type **attd** to send all calls to the attendant.

   This is the phone extension that receives the referral call when a security violation occurs.

5. In the Login Threshold field, type **3**.

   This is the minimum number of login attempts that are permitted before a referral call is made. More than 3 attempts causes a security violation notification.

6. In the Time Interval field, type **0:03**.

   This the time interval in which the threshold, or number of violations, must occur.

7. Press ENTER to save your changes.

   NOTE:
   The following 3 steps are optional. If you are not using Remote Access, go to step 11.

8. Type **change remote-access** and press RETURN.

   The QSIG to DCS TSC Gateway screen screen appears.

```
                                  REMOTE ACCESS
   Remote Access Extension________           Barrier Code Length____
Authorization Code Required? y                    Remote Access Dial Tone: n
Barrier Code   COR  TN  COS    Expiration Date    No. of Calls     Calls Used
1:________     1__  1_  1__    __/__/__           ______           ______
2:________     1__  1_  1__    __/__/__           ______           ______
3:________     1__  1_  1__    __/__/__           ______           ______
4:________     1__  1_  1__    __/__/__           ______           ______
5:________     1__  1_  1__    __/__/__           ______           ______
6:________     1__  1_  1__    __/__/__           ______           ______
7:________     1__  1_  1__    __/__/__           ______           ______
8:________     1__  1_  1__    __/__/__           ______           ______
9:________     1__  1_  1__    __/__/__           ______           ______
10:_______     1__  1_  1__    __/__/__           ______           ______
Permanently Disable? __  Disable Following A Security Violation? y
(NOTE:  You must logoff to effect permanent disabling of Remote Access)
```

9. In the Disable Following A Security Violation field, type **y**.

   This disables Remote Access following detection of a remote access security violation.

10. Press ENTER to save your changes.

11. Type **change station xxxx** and press RETURN, where xxxx is the station to be assigned the notification halt button.

    The Station screen appears.

```
                                                       Page 3 of X
                                      STATION
     SITE DATA
          Room: _______                                   Headset? n
          Jack: ___                                       Speaker? n
         Cable: ___                                      Mounting: d
         Floor: _______                              Cord Length: 0_
      Building: _______                                Set Color: _____


ABBREVIATED DIALING
     List1: ________          List2: _________          List3: _________

BUTTON ASSIGNMENTS
 1: asvn-halt                         4: ________
 2: ________                          5: ________
 3: ________
```

12. In the Feature Button Assignments section, type one of the following:
    - **asvn-halt** — The Authorization Code Security Violation Notification call is activated when an authorization code security violation is detected. This applies only if you are using authorization codes.
    - **lsvn-halt** — The Login Security Violation Notification call is activated a referral call when a login security violation is detected.
    - **rsvn-halt** — The Remote Access Barrier Code Security Violation Notification call is activated as a call referral. This applies only if you are using Remote Access barrier codes.
    - **ssvn-halt** — The Station Code Security Violation Notification call is activated when a station code security violation is detected. This applies only if you are using station codes.

    NOTE:
    Any of the above 4 security violations will cause the system to place a notification call to the designated phone. The call continues to ring until answered. To stop notification of any further violations, press the button associated with the type of violation.

13. Press ENTER to save your changes.

Setting up authorization codes

Authorization codes provide the means for extending control of system users' calling privileges. They extend calling-privilege control and provide an extra level of security for remote-access callers.

NOTE:
To maintain system security, Avaya recommends you use authorization codes.

Refer to *Avaya Products Security Handbook* for more information.

Before you start

On the System-Parameters Customer-Options screen, verify the Authorization Codes field is **y**. If not, contact your Avaya representative. This field turns on the feature and permits you to selectively specify levels of calling privileges that override in-place restrictions.

Instructions

1. Type **change system-parameters features** and press RETURN.

   The Feature-Related System Parameters screen appears.

```
                                                                Page 3 of 8
                       FEATURE-RELATED SYSTEM PARAMETERS

                        Reserved Slots for Attendant Priority Queue: 5_
                                         Time Before Off-Hook Alert: 10__
                             Emergency Access Redirection Extension: _____
               Number of Emergency Calls Allowed in Attendant Queue: __
                                               Call Pickup Alerting? n
                        Temporary Bridged Appearance on Call Pickup? y
                                      Call Pickup on Intercom Calls? y
                                               Directed Call Pickup? n
                  Deluxe Paging and Call Park Timeout to Originator? n
Controlled Outward Toll Restriction Intercept Treatment: tone _______
    Controlled Termination Restriction (Do Not Disturb): tone _______
              Controlled Station to Station Restriction: tone _______
AUTHORIZATION CODE PARAMETERS            Authorization Code Enabled? y
                                          Authorization Code Length: 7
                             Authorization Code Cancellation Symbol? #
                                            Attendant Time Out Flag? n
                                         Display Authorization Code? _
                       Controlled Toll Restriction Replaces: station-station
          Controlled Toll Restriction Intercept Treatment: extension     3000
```

2. In the Authorization Code Enabled field, type **y**.

   This enables the Authorization Codes feature on a system-wide basis.

3. In the Authorization Code Length field, type **7.**

   This defines the length of the Authorization Codes your users need to enter. To maximize the security of your system, Avaya recommends you make each authorization code the maximum length allowed by the system.

4. In the Authorization Code Cancellation Symbol field, leave the default of **#**.

   This is the symbol a caller must dial to cancel the 10-second wait period during which your user can enter an authorization code.

5. In the Attendant Time Out Flag field, leave the default of **n**.

   This means a call is not to be routed to the attendant if a caller does not dial an authorization code within 10 seconds or dials an invalid authorization code.

6. In the Display Authorization Code field, type **n**.

   This prevents the authorization code from displaying on phone sets thus maximizing your security.

7. Press ENTER to save your changes.

8. Type **change authorization-code *nnnn*** and press RETURN, where *nnnn* is the authorization code.

   The Authorization Code — COR Mapping screen appears.

```
                                                           Page  1  of  1
                Authorization Code - COR Mapping
   Note:     XX codes administered.  Use "list" to display all codes.
  AC     COR    AC     COR    AC     COR    AC     COR    AC      COR    AC     COR
 4285193 1_   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
 _______ __   _______ __   _______ __   _______ __   _______ __   _______ __
```

9. In the AC field, enter the authorization code your users must dial.

   In our example, type **4285193**. The number of digits entered must agree with the number assigned in the Feature-Related System Parameters screen, Authorization Code Length field.

   NOTE:
   Remember, all authorization codes used in the system must be the same length.

10. In the COR field, enter the desired Class of Restriction number from 0 through 95.

    In our example, type **1**.

11. Press ENTER to save your changes.

Related topics

Refer to "Class of Restriction" on page 636 for more information on setting up dialing out restrictions.

Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information on using trunk access codes.

Refer to ''Facility restriction levels and traveling class marks'' on page 1564 and ''Route Pattern'' on page 1039 for more information on assigning Facility Restriction Levels.

Refer to ''Call Detail Recording'' on page 1437 and ''Station'' on page 1066 for more information on using Call Detail Recording on station phones.

Refer to ''Class of Restriction'' on page 636 and ''Station'' on page 1066 for more information on using Class of Restriction on station phones.

Refer to ''Remote Access'' on page 1680 for more information on allowing authorized callers to access the system from remote locations.

Refer to''Barrier codes'' on page 1388 or ''QSIG to DCS TSC Gateway screen'' on page 1029 for information on barrier codes.

Dealing with security violations

When a security violation occurs, there are steps that you can take to be sure that this same attempt is not successful in the future.

Disabling a login ID

There may be occasions when you have to disable a login for one of your users because of a security violation.

1. Log in to the switch using a login ID with the correct permissions.
2. Type **disable login ge0rg3** and press RETURN.

Enabling a login ID

You may have to enable a login ID that has been disabled by a security violation, or disabled manually with the disable login command.

1. Log in to the switch using a login ID with the correct permissions.
2. Type **enable login ge0rg3** and press RETURN.

Enabling remote access

You may have to enable Remote Access that has been disabled following a security violation, or disabled manually.

1. Log in to the switch using a login ID with the correct permissions.
2. Type **enable remote-access** and press RETURN.

Disabling remote access

There may be occasions when you have to disable remote access for one of your users because of a security violation.

1. Log in to the switch using a login ID with the correct permissions.
2. Type **disable remote-access** and press RETURN.

Managing trunks

12

This chapter contains basic procedures for working with analog and digital trunks. Specialized trunks such as Internet Protocol (IP), H.323, APLT, tandem, release-link, and DMI-BOS trunks are not covered in this manual. For more information, refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software*. This chapter also does not contain procedures for working with ISDN trunk groups. Due to the complexity of ISDN technology and the potential consequences of errors, ask your Avaya representative to assist you in planning, installing, and administering ISDN trunks. For an introduction to ISDN service on Avaya MultiVantage, refer to ‘‘ISDN service’’ on page 1610. For descriptions of the fields on various screens used to set up H.323 trunks, see Chapter 18, ‘‘Screen reference’’.

Tips for working with trunk groups

You’ll find detailed procedures for administering specific trunk groups elsewhere in this chapter. However, there’s more to working with trunks than just administering trunk groups.

Following a process

Trunking technology is complex. Following a process can prevent mistakes and save you time. To set up new trunks and trunk groups, Avaya recommends following the process below (some steps may not apply to your situation):

1. Install the necessary circuit packs and perform any administration the circuit pack requires.
2. Connect the appropriate ports to your network service provider’s trunks.

3. Administer a trunk group to control the operation of the trunks.
4. Assign the ports you're using to the trunk group.
5. For outgoing or 2-way trunks, administer Automatic Route Selection so Avaya MultiVantage knows which outgoing calls to route over this trunk group.
6. Test your new trunk group. Using the trunk access code, place a variety of calls.

This chapter provides instructions for steps 3 and 4 in this process.

Working with your network service provider

Depending on the type of trunk you want to add, the vendor may be your local phone company, a long distance provider, or some other service provider. Key settings on Avaya MultiVantage must be identical to the same settings on the provider's equipment for your trunks to work. Clear, frequent communication with your provider is essential — especially since some providers may use different terms and acronyms than Avaya does!

Once you decide that you want to add a new trunk, contact your vendor. The vendor should confirm the type of signal you want and provide you with a circuit identification number for the new trunk. Be sure to record any vendor-specific ID numbers or specifications in case you ever have any problems with this trunk.

Keeping records

In addition to recording vendor-specific information such as ID numbers, you should record the following information about every trunk group you have.

The questions you need to answer	The kind of information you need to get
What type of trunk group is it?	You need to know what kind of trunks these are (central office (CO), foreign exchange (FX), etc.) and whether they use any special services (such as T1 digital service). You also need to know what kind of signaling the group uses. For example, you might have a CO trunk group with ground-start signaling running on a robbed-bit T1 service.

The questions you need to answer	The kind of information you need to get
Which phone numbers are associated with each trunk group?	For incoming or two-way trunk groups: ■ What number or numbers do outside callers use to call into your switch over this group? ■ What's the destination extension to which this trunk group delivers calls? Does it terminate at an attendant or a voice-mail system? For outgoing trunk groups: ■ What extensions can call out over this trunk group?
Is the service from your network service provider sending digits on incoming calls?	Direct Inward Dial and Direct Inward/Outward Dial trunks send digits to your switch. Tie trunks may send digits, depending on how they're administered. You need to know: ■ How many digits is your service provider sending? ■ Are you inserting any digits? What are they? ■ Are you absorbing any digits? How many? ■ What range of numbers has your service provider assigned you?

Helpful tips for setting common fields

The procedures in this section cover the specific fields you must administer when you create each type of trunk group. Here are some tips for working with common fields that are available for most trunk groups.

- Dial Access — Typing **y** in this field allows users to route calls through an outgoing or two-way trunk group by dialing its trunk access code.

  SECURITY ALERT:
  Calls dialed with a trunk access code over Wide Area Telecommunications Service (WATS) trunks are not validated against the ARS Digit Analysis Table, so users can dial anything they wish. For security, you may want to leave the field set to ***n*** *unless you need dial access to test the trunk group.*

- Outgoing Display — Typing **y** in this field allows display phones to show the name and group number of the trunk group used for an outgoing call. This information may be useful to you when you're trying to diagnose trunking problems.
- Queue Length — Don't create a queue for two-way loop-start trunks, or you may have a problem with glare (the interference that happens when a two-way trunk is seized simultaneously at both ends).
- Trunk Type — Use ground-start signaling for two-way trunks whenever possible: ground-start signaling avoids glare and provides answer supervision from the far end. Try to use loop-start signaling only for one-way trunks.

Related topics

Refer to the *Avaya MultiVantage™ Hardware Solutions Guide* for information on the types of circuit packs available and their capacities.

Refer to your switch's installation manual for installation instructions for circuit packs.

Refer to ''Routing outgoing calls'' on page 243 for detailed information on Automatic Route Selection.

Adding a CO, FX, or WATS trunk group

Basic administration for Central Office (CO), Foreign Exchange (FX), and Wide Area Telecommunication Service (WATS) trunk groups is identical, so we've combined instructions for all 3 in the following procedure. In most cases, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information. Your settings in the following fields *must* match your provider's settings:

- Direction
- Comm Type
- Trunk Type

CAUTION:
Use the list above as a starting point and talk to your service provider. Depending on your particular application, you may need to coordinate additional administration with your service provider.

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, refer to the *Avaya MultiVantage™ Hardware Solutions Guide.*

Instructions

As an example, let’s set up a two-way CO trunk group that carries voice and voice-grade data only. Incoming calls terminate to an attendant during business hours and to a night service destination the rest of the time

To add the new CO trunk-group:

1. Type **add trunk-group next** and press RETURN.

   The Trunk Group screen appears. The system assigns the next available trunk group number to this group. In our example, we’re adding trunk group 5.

```
                               TRUNK GROUP

Group Number: 5                   Group Type: co          CDR Reports: y
  Group Name: Outside calls              COR: 85       TN: 1__       TAC: 105
   Direction: two-way        Outgoing Display? n
 Dial Access? n                Busy Threshold: 99     Night Service: 1234____
Queue Length: 0           Country:             Incoming Destination: attd____
   Comm Type: voice                    Auth Code? n  Digit Absorption List: _
    Prefix-1? y                      Trunk Flash? n        Toll Restricted? y

TRUNK PARAMETERS
            Trunk Type: ground start
    Outgoing Dial Type: tone                                 Cut-Through? n
     Trunk Termination: rc                    Disconnect Timing(msec): 500_

            Auto Guard? n    Call Still Held? n     Sig Bit Inversion: none

                                  Trunk Gain: high

Disconnect Supervision -In? y  Out? n
Answer Supervision Timeout: 10            Receive Answer Supervision? n
```

2. In the Group Type field, type **co**.

   This field specifies the kind of trunk group you’re creating.

3. In the Group Name field, type **Outside calls**.

   This name will be displayed, along with the group number, for outgoing calls if you set the Outgoing Display? field to y. You can type any name up to 27 characters long in this field.

4. Type **85** in the COR field.

   This field controls which users can make and receive calls over this trunk group. Assign a class of restriction that's appropriate for the COR calling permissions administered on your system.

5. In the TAC field, type **105**.

   This field defines a unique code that you or your users can dial to access this trunk group. The code also identifies this trunk group in call detail reports.

6. In the Direction field, type **two-way**.

   This field defines the direction of traffic flow on this trunk group.

7. Type **1234** in the Night Service field.

   This field assigns an extension to which calls are routed outside of business hours.

8. In the Incoming Destination field, type **attd**.

   This field assigns an extension to which incoming calls are routed during business hours. By entering attd in this field, incoming calls go to the attendant and the system treats the calls as Listed Directory Number calls.

9. In the Comm Type field, type **voice**.

   This field defines whether a trunk group can carry voice, data, or both. Analog trunks only carry voice and voice-grade data.

10. In the Trunk Type field, type **ground-start**.

    This field tells the system what kind of signaling to use on this trunk group. To prevent glare, ground start signaling is recommended for most two-way CO, FX, and WATS trunk groups.

11. In the Outgoing Dial Type field, type **tone**.

    This field tells the switch how digits are to be transmitted for outgoing calls. Entering tone actually allows the trunk group to support both DTMF and rotary signals, so Avaya recommends that you always put tone in this field.

12. In the Trunk Termination field, type **rc**.

    Use rc in this field when the distance to the central office or the switch at the other end of the trunk is more than 3,000 feet. Check with your service provider if you're not sure of the distance to your central office.

13. Press ENTER to save your changes.

Now you're ready to add trunks to this trunk group. Refer to ''Adding trunks to a trunk group'' on page 422.

Adding a DID trunk group

In most cases, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information. For Direct Inward Dialing (DID) trunk groups, settings in the following fields *must* match your provider's settings:

- Direction
- Comm Type
- Trunk Type
- Expected Digits (only if the digits your provider sends *do not* match your dial plan)

CAUTION:

Use the list above as a starting point and talk to your service provider. Depending on your particular application, you may need to coordinate additional administration with your service provider.

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, refer to the *Avaya MultiVantage™ Hardware Solutions Guide.*

Tip:

In the DID/Tie/ISDN Intercept Treatment field on the Feature-Related System parameters screen, enter **attd**. *Incoming calls to invalid extensions will be routed to the attendant.*

Instructions

To add the new Direct Inward Dialing (DID) trunk-group:

1. Type **add trunk-group next** and press RETURN.

   The Trunk Group screen appears. The system assigns the next available trunk group number to this group. In our example, we're adding trunk group 5.

```
                              TRUNK GROUP

Group Number: 5                Group Type: did            CDR Reports: y
  Group Name: Incoming calls          COR: 85          TN: 1       TAC: 105

                     Country: 1
                                Auth Code? n


TRUNK PARAMETERS
         Trunk Type: wink-start   Incoming Rotary Timeout(sec): 5
                                            Incoming Dial Type: tone
  Trunk Termination: rc              Disconnect Timing(msec): 500
    Digit Treatment: insertion                         Digits: 6
    Expected Digits: 4                         Sig Bit Inversion: none
  Analog Loss Group: ___                       Digital Loss Group: ___
Extended Loop Range? n      Trunk Gain: high    Drop Treatment: silence

 Disconnect Supervision - In? y
```

2. In the Group Type field, type **did**.

   This field specifies the kind of trunk group you're creating.

3. In the Group Name field, type **Incoming calls**.

   You can type any name up to 27 characters long in this field.

4. Type **85** in the COR field.

   This field controls which users can receive calls over this trunk group. Assign a class of restriction that's appropriate for the COR calling permissions administered on your system.

5. In the TAC field, type **105**.

   This code identifies the trunk group on CDR reports.

6. In the Trunk Type field, type **wink-start**.

   This field tells the system what kind of signaling to use on this trunk group. In most situations, use wink start for DID trunks to minimize the chance of losing any of the incoming digit string.

7. In the Incoming Dial Type field, type **tone**.

   This field tells the switch how digits are transmitted for incoming calls. Entering tone actually allows the trunk group to support both DTMF and rotary signals, so Avaya recommends that you always put tone in this field.

8. In the Trunk Termination field, type **rc**.

   Use rc in this field when the distance to the central office or the switch at the other end of the trunk is more than 3,000 feet. Check with your service provider if you're not sure of the distance to your central office.

9. Press ENTER to save your changes.

Now you're ready to add trunks to this trunk group. Refer to ''Adding trunks to a trunk group'' on page 422.

Related topics

Refer to ''Inserting and absorbing digits'' on page 426 for instructions on matching modifying incoming digit strings to match your dial plan.

Adding a DIOD trunk group

Administration for Direct Inward and Outward Dialing (DIOD) trunk groups varies from country to country. Refer to *Avaya MultiVantage™ Application Notes for Type Approval*, and use the DS1 screen illustrations in the chapter for your country as a model for your administration. Remember that the central office serving your switch may be emulating another country's network protocol. If so, you'll have to administer your circuit packs and trunk groups to match the protocol used by your central office.

SECURITY ALERT:

If you're using Incoming Caller ID (ICLID) on analog trunks connected to a DIOD Central Office trunk circuit pack, DO NOT put these trunks in an outgoing AAR or ARS route pattern. Since the loop-start trunks supported on the DIOD Central Office trunk circuit pack do not provide answer supervision, the potential for toll fraud exists.

Adding a PCOL trunk group

In most cases, when administering Personal Central Office Line (PCOL) trunk groups, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information. Your settings in the following fields *must* match your provider's settings:

- Trunk Type
- Trunk Direction

CAUTION:

Use the list above as a starting point and talk to your service provider. Depending on your particular application, you may need to coordinate additional administration with your service provider.

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, refer to the *Avaya MultiVantage™ Hardware Solutions Guide.*

Instructions

As an example, let's set up a new PCOL group and administer the group as a CO trunk for two-way voice traffic.

To add the new PCOL group:

1. Type **add personal-co-line next** and press ENTER.

The Personal CO Line Group screen appears.

```
                                                                    Page 1 of X
                              PERSONAL CO LINE GROUP
 Group Number: 1                     Group Type: co            CDR Reports: y
   Group Name: OUTSIDE CALL                                            TAC: ____
Security Code: ___                Coverage Path: ____     Data Restriction? _
                               Outgoing Display? _

TRUNK PARAMETERS
                 Trunk Type: ground-start           Trunk Direction: two-way
                 Trunk Port: 01D1901_       Disconnect Timing(msec): 500
                 Trunk Name: __________           Trunk Termination: rc____
        Outgoing Dial Type: tone_____            Analog Loss Group: 7
                   Prefix-1? y                  Digital Loss Group: _
 Answer Supervision Timeout: ___         Receive Answer Supervision? _
                 Trunk Gain: high                           Country: 1
          Charge Conversion: 1____
              Decimal Point: none__
            Currency Symbol: ___
                Charge Type: units__
```

2. In the Group Type field, type **co**.

   This field specifies the kind of trunk group you're creating. PCOL groups can be administered as CO, FX, or WATS trunks.

3. In the Group Name field, type **Outside calls**.

   This name will be displayed, along with the group number, for outgoing calls if you set the Outgoing Display? field to **y**. You can type any name up to 27 characters long in this field. (You may want to put the phone number here that's assigned to this trunk.)

4. In the TAC field, type **111**.

   This field defines a unique code that you or your users can dial to access this trunk group. The code also identifies this trunk group in call detail reports.

5. In the Trunk Type field, type **ground start**.

   This field tells the system what kind of signaling to use on this trunk group. To prevent glare, ground start signaling is recommended for most two-way CO, FX, and WATS trunk groups.

6. In the Trunk Port field, type **01D1901**.

   This is the port to which the trunk is connected.

7. In the Trunk Termination field, type **rc**.

   Use rc in this field when the distance to the central office or the switch at the other end of the trunk is more than 3,000 feet. Check with your service provider if you're not sure of the distance to your central office.

8. In the Outgoing Dial Type field, type **tone**.

   This field tells the switch how digits are to be transmitted for outgoing calls. Entering tone actually allows the trunk group to support both DTMF and rotary signals, so Avaya recommends that you always put tone in this field.

9. Press ENTER to save your changes.

You assign phones to a PCOL group by administering a CO Line button on each phone. Once assigned, the Assigned Members page of the Personal CO Line Group screen displays member phones:

```
                                                           Page 2 of X
                          PERSONAL CO LINE GROUP

ASSIGNED MEMBERS (Stations with a button for this PCOL Group)

      Ext    Name                           Ext    Name

    1:                                   9:
    2:                                  10:
    3:                                  11:
    4:                                  12:
    5:                                  13:
    6:                                  14:
    7:                                  15:
    8:                                  16:
```

More information

Call Detail Recording

Call Detail Recording can be activated for calls on a personal CO line, but the CDR record does not specifically identify the call as PCOL. Calls over personal CO lines can, however, be identified by the trunk access code used on the call. The call is recorded to the extension number assigned to the phone where the call was originated or answered.

Restrictions

- Abbreviated Dialing can be used with a personal CO line, but the accessed lists are associated with the individual phones.
- Auto Hold and Leave Word Calling do not work with calls on a personal CO line.
- Send All Calls cannot be activated for a personal CO line.
- INTUITY AUDIX cannot be in the coverage path of a PCOL group.

- Only phones in the same PCOL group can bridge onto calls on the personal CO line. If a user is active on his or her primary extension number on a PCOL call, bridged call appearances of that extension number cannot be used to bridge onto the call.
- When a user puts a call on hold on a personal CO line, the status lamp associated with the PCOL button does not track the busy/idle status of the line.

Adding a Tie or Access trunk group

In most cases, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information. Your settings in the following fields *must* match your provider's settings (or the setting on the far-end switch, if this is a private network trunk group):

- Direction
- Comm Type
- Trunk Type

CAUTION:

Use the list above as a starting point and talk to your service provider. Depending on your particular application, you may need to coordinate additional administration with your service provider.

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, refer to the *Avaya MultiVantage™ Hardware Solutions Guide.*

Tip:

In the DID/Tie/ISDN Intercept Treatment field on the Feature-Related System parameters screen, enter **attd**. *Incoming calls to invalid extensions will be routed to the attendant.*

Instructions

As an example, let's add a two-way tie trunk group that supports voice and voice-grade data.

To add the new tie trunk-group:

1. Type **add trunk-group next** and press RETURN.

   The Trunk Group screen appears. The system assigns the next available trunk group number to this group. In our example, we're adding trunk group 5.

```
                                                          Page 1 of X
                                TRUNK GROUP

Group Number: 5__                  Group Type: tie          CDR Reports: y
  Group Name: Outside calls               COR: 85_      TN: 1__      TAC: 105_
   Direction: two-way_      Outgoing Display? n    Trunk Signaling Type: ____
 Dial Access? n               Busy Threshold: 99_         Night Service: 1234
Queue Length: 0__                                Incoming Destination: _______
   Comm Type: voice                 Auth Code? n
                                  Trunk Flash? n

         BCC: 0
TRUNK PARAMETERS
   Trunk Type (in/out): wink/wink___     Incoming Rotary Timeout(sec): 5_
    Outgoing Dial Type: tone_____                  Incoming Dial Type: tone____
                                              Disconnect Timing(msec): 500_
       Digit Treatment: __________                             Digits: ____
       Expected Digits:__                          Sig Bit Inversion: none
     Analog Loss Group: ___                        Digital Loss Group: ___
    Incoming Dial Tone? y

 Disconnect Supervision - In? y  Out? n
  Answer Supervision Timeout: 0__        Receive Answer Supervision? y
```

2. In the Group Type field, type **tie**.

   This field specifies the kind of trunk group you're creating.

3. In the Group Name field, type **Outside calls**.

   This name will be displayed, along with the group number, for outgoing calls if you set the Outgoing Display? field to y. You can type any name up to 27 characters long in this field.

4. Type **85** in the COR field.

   This field controls which users can make or receive calls over this trunk group. Assign a class of restriction that's appropriate for the COR calling permissions administered on your system.

5. In the TAC field, type **105**.

   This field defines a unique code users can dial to access this trunk group.

6. In the Direction field, type **two-way**.

   This field defines the direction of traffic flow on this trunk group.

7. Type **1234** in the Night Service field.

   This field assigns an extension to which calls are routed outside of business hours.

8. In the Comm Type field, type **voice**.

   This field defines whether a trunk group can carry voice, data, or both. Analog trunks only carry voice and voice-grade data. If you're administering a T1 connection in North America, type **rbavd** in this field.

9. In the Trunk Type field, type **wink/wink**.

   This field tells the system what kind of signaling to use on this trunk group. Because we're receiving and sending digits over this trunk group, we're using wink/wink signaling to minimize the chance of losing part of the digit string in either direction.

10. Type **tone** in both the Outgoing Dial Type and Incoming Dial Type fields.

    These fields tell the switch how digits are transmitted for incoming calls. Entering tone actually allows the trunk group to support both DTMF and rotary signals, so Avaya recommends that you always put tone in this field.

11. Press ENTER to save your changes.

Now you're ready to add trunks to this trunk group. Refer to ‘‘Adding trunks to a trunk group’’ on page 422.

Setting up digital trunks

Any of the common trunks, except for PCOL trunks, can be analog or digital. (PCOL trunks can only be analog.) Administering a digital trunk group is very similar to administering its analog counterpart, but digital trunks must connect to a DS1 circuit pack and this circuit pack must be administered separately. The example in this section shows you how to do this.

In most cases, Avaya recommends leaving the default settings in fields that aren't specifically mentioned in the following instructions. Your Avaya representative or network service provider can give you more information.

Your settings in the following fields *must* match your provider's settings:

- Bit Rate
- Line Coding (unless you're using a channel service unit to convert between your line coding method and your provider's)
- Framing Mode

- Signaling Mode
- Interface Companding

CAUTION:
Use the list above as a starting point and talk to your service provider. Depending on your particular application, you may need to coordinate additional administration with your service provider.

Before you start

Assign the DS1 circuit pack before you administer the members of the associated trunk groups.

CAUTION:
If enhanced DS1 administration is not enabled, you cannot make changes to the DS1 Circuit Pack screen before you remove related member translations of all trunks from the trunk group. Refer to *“Enhanced DS1 administration’’ on page 420*.

Before you can administer a digital trunk group, you must have one or more circuit packs that support DS1 with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, refer to the *Avaya MultiVantage™ Hardware Solutions Guide.*

Instructions

The following example shows a DS1 circuit pack configured for T1 service. The circuit pack is supporting a two-way CO trunk group that carries only voice and voice-grade data.

To configure a new DS1 circuit pack:

1. Type **add ds1 *07A19*** and press ENTER.

   The DS1 Circuit Pack screen appears. You must enter a specific port address for the circuit pack.

```
                                  DS1 CIRCUIT PACK

                 Location: 07A19                            Name: two-way CO
                 Bit Rate: 1.544                     Line Coding: b8zs
        Line Compensation: 1                        Framing Mode: esf
           Signaling Mode: robbed-bit


     Interface Companding: mulaw
                Idle Code: 11111111



           Slip Detection? y
```

2. Type **two-way CO** in the Name field.

   Use this name to record useful information such as the type of trunk group associated with this circuit pack or its destination.

3. In the Bit Rate field, type **1.544**.

   This is the standard for T1 lines.

4. In the Line Coding field, type **b8zs**.

   Avaya recommends you use b8zs whenever your service provider supports it. Since this trunk group only carries voice traffic, you could also use ami-zcs without a problem.

5. In the Framing Mode field, type **esf**.

   Avaya recommends you use esf whenever your service provider supports it.

6. In the Signaling Mode field, type **robbed-bit**.

7. In the Interface Companding field, type **mulaw**.

   This is the standard for T1 lines in North America.

8. Press ENTER to save your changes.

More information

T1 recommended settings

The table below shows recommended settings for standard T1 connections to your local exchange carrier.

Field	Value	Notes
Line Coding	b8zs	Use ami-zcs if b8zs is not available.
Signaling Mode	robbed-bit	Robbed-bit signaling gives you 56K bandwidth per channel. If you need a 64K clear channel for applications like asynchronous data transmission or remote administration access, use common channel signaling.
Framing	esf	Use d4 if esf is not available.

If you use b8zs line coding and esf framing, it will be easier to upgrade your T1 facility to ISDN should you want to. You can upgrade without reconfiguring external channel service units, and your service provider won't have to reconfigure your network connection.

E1 recommended settings

DS1 administration for E1 service varies from country to country. Refer to *Avaya MultiVantage Application Notes for Type Approval* and use the DS1 screen illustrations in the chapter for your country as a model for your administration.

NOTE:
Remember that the central office serving your switch may be emulating another country's network protocol. If so, you'll have to administer your circuit packs and trunk groups to match the protocol used by your central office.

Enhanced DS1 administration

Normally, you can't change the DS1 Circuit Pack screen unless you remove all related trunks from their trunk group. However, if the DS1 MSP field on the System-Parameters Customer-Options screen is **y** and you are assigned the associated login permissions, you can change some of the fields on the DS1 Circuit Pack screen without removing the related trunks from their trunk group.

The following enhanced DS1 administration login permissions must be assigned on the Command Permission Categories screen:

- The Maintain Enhanced DS1 field must be **y**.
- The Maintain Trunks field must be **y**.
- The Maintain Switch Circuit Packs field must be **y**.

If you busy out the DS1 circuit pack, you can change the following fields: CRC, Connect, Country Protocol, Framing Mode, Interface, Interconnect, Line Coding, and Protocol Version. After changing these fields, you may also have to change and resubmit associated screens.

Matching field settings on different screens

For enhanced DS1 administration, some field values on the DS1 Circuit Pack screen must be consistent with those on other screens as shown in the table below. If you change field values on the DS1 screen, you must change the related fields on the other screens and resubmit them.

DS1 Circuit Pack field	Affected screens
Line Coding	Route Pattern Access Endpoint PRI Endpoint Signaling Group Trunk Group
Connect	Signaling Group
Protocol Version	Signaling Group
Interface	Signaling Group
Interconnect	Trunk Group
Country Protocol	Signaling Group Trunk Group

Specific combinations of settings for some of these fields are shown below.

ITC, Bit Rate, and Line Coding. The ITC (Information Transfer Capability) field appears on the Route Pattern screen, Trunk Group screen, and Access Endpoint screen. The Line Coding and the Bit Rate fields appear on the DS1 Circuit Pack screen. The settings for these fields on all the screens must be coordinated as shown in the following tables.

ITC field	Bit Rate	Line Coding field
restricted	1.544 Mbps	ami-zcs
	2.048 Mbps	ami-basic
unrestricted	1.544 Mbps	b8zs
	2.048 Mbps	hdb3

Interconnect and corresponding Group Type entries. The Interconnect field appears on the DS1 Circuit Pack screen. The Group Type field appears on the Trunk Group screen. Set these fields as shown in the following table.

Interconnect field	Group Type field
co	co, did, diod, fx, or wats
pbx	access, aplt, isdn-pri, tandem, or tie

Related topics

Refer to “DS1 Circuit Pack” on page 728 for information on administering DS1 service.

Refer to “DS1 Trunk Service” on page 1542 for detailed information on DS1 service.

Adding trunks to a trunk group

Use this procedure to add new trunks or to change the assignment of existing trunks. To change the assignment of existing trunks, remove them from their current trunk group and add them to the new group.

Before you start

You must add a trunk group before you can assign and administer individual trunks. To add a new trunk group, refer to the instructions in this chapter for the type of group you want to add.

Instructions

As an example, let’s assign 5 trunks to a new tie trunk group, trunk group 5. We’ll use ports on several circuit packs for members of this group.

To assign trunks to trunk group 5:

1. Type **change trunk-group 5** and press RETURN.

   The Trunk Group screen appears.

2. Move to the Group Member Assignments page.

   Some of the fields on this screen won’t appear for every trunk group.

```
                                  TRUNK GROUP
                                         Administered Members(min/max): xxx/yyy
                                            Total Administered Members: xxx
GROUP MEMBER ASSIGNMENTS
       Port     Code   Sfx  Name          Night            Mode      Type    Ans Delay
   1: 1B1501_  TN464   F   5211_______   _____           e&m_____  t1-comp  ____
   2: 1B1502_  TN464   F   5212_______   _____           e&m_____  t1-comp  ____
   3: 1B1601_  TN464   F   5213_______   _____           e&m_____  t1-comp  ____
   4: 1B1602_  TN464   F   5214_______   _____           e&m_____  t1-comp  ____
   5: 1B1701_  TN464   F   5215_______   _____           e&m_____  t1-comp  ____
   6: _______               ___________  _____           ________  _______  ____
   7: _______               ___________  _____           ________  _______  ____
   8: _______               ___________  _____           ________  _______  ____
   9: _______               ___________  _____           ________  _______  ____
  10: _______               ___________  _____           ________  _______  ____
  11: _______               ___________  _____           ________  _______  ____
  12: _______               ___________  _____           ________  _______  ____
  13: _______               ___________  _____           ________  _______  ____
  14: _______               ___________  _____           ________  _______  ____
  15: _______               ___________  _____           ________  _______  ____
```

3. In the Port field in row 1, type **1B1501**.

   This field assigns the first member of the trunk group to a port on a circuit pack.

4. In the Name field in row 1, type **5211**.

   This is the extension assigned to this trunk. In general, type the circuit ID or telephone number for each trunk in this field. The information is helpful for tracking your system or troubleshooting problems. Update these fields whenever the information changes.

5. In the Mode field, type **e&m**.

   CAUTION:
   An entry in this field is only required for some circuit packs. Dip switch settings on the circuit pack control the signalling mode used on the trunk group, so the entry in the Mode field must correspond to the actual setting on the circuit pack.

6. In the Type field, type **t1-comp**.

   An entry in this field is only required for some circuit packs.

7. Repeat steps 3–6, as appropriate, for the remaining trunks.

   Notice that you can assign trunks in the same trunk group to ports on different circuit packs.

8. Press ENTER to save your changes.

Removing trunks from a trunk group

Use this procedure to remove vacant trunk ports or to change the assignment of existing trunks. To change the assignment of existing trunks, remove them from their current trunk group and add them to the new group.

Instructions

Let's remove trunks 5214 and 5215 from trunk group 5.

To remove trunks from trunk group 5:

1. Type **change trunk-group 5** and press RETURN.

   The Trunk Group screen appears.

```
                                  TRUNK GROUP
                                       Administered Members(min/max):   3/5
                                          Total Administered Members:   5
GROUP MEMBER ASSIGNMENTS
      Port     Code  Sfx  Name         Night          Mode     Type     Ans Delay
   1: 1B1501_ TN464   F   5211_______ _____          e&m_____ t1-comp ____
   2: 1B1502_ TN464   F   5212_______ _____          e&m_____ t1-comp ____
   3: 1B1601_ TN464   F   5213_______ _____          e&m_____ t1-comp ____
   4: 1B1602_ TN464   F   5214_______ _____          e&m_____ t1-comp ____
   5: 1B1701_ TN464   F   5215_______ _____          e&m_____ t1-comp ____
   6: _______                 ___________ _____          ________ _______ ____
   7: _______                 ___________ _____          ________ _______ ____
   8: _______                 ___________ _____          ________ _______ ____
   9: _______                 ___________ _____          ________ _______ ____
  10: _______                 ___________ _____          ________ _______ ____
  11: _______                 ___________ _____          ________ _______ ____
  12: _______                 ___________ _____          ________ _______ ____
  13: _______                 ___________ _____          ________ _______ ____
  14: _______                 ___________ _____          ________ _______ ____
  15: _______                 ___________ _____          ________ _______ ____
```

2. Move to the Group Member Assignments page.

   Some of the fields on this screen won't appear for every trunk group.

3. Move the cursor to the Port 4 field and clear its entry.

4. Move the cursor to the Port 5 field and clear its entry.
5. Press ENTER to save your changes.

   The switch will automatically clear the other fields associated with these 2 trunks.

Removing trunk groups

There's more to removing a trunk group than just executing the remove trunk-group command. If you're using ARS, you must remove an outgoing or two-way trunk group from any route patterns that use it. If you've administered Trunk-Group Night Service buttons for the trunk group on any phones, those buttons must be removed or assigned to another trunk group.

Instructions

As an example, let's remove trunk group 5. This two-way group is used in ARS route pattern 2. In addition, a Trunk-Group Night Service button on extension 8410 points to this group.

To remove trunk group 5:

1. In the Route Pattern screen for route pattern 2, clear the entries for trunk group 5.

   If you're replacing trunk group 5 with another trunk group, just type the information for the new trunk group over the old entries. Remember to press ENTER to save your changes.
2. In the Station screen for extension 8410, clear the entry in the Button Assignments field for the Trunk-Group Night Service button.

   Remember to press ENTER to save your changes.
3. In the Trunk Group screen for trunk group 5, remove all member trunks from the group.

   Refer to ''Removing trunks from a trunk group'' on page 424 for instructions.
4. Type **remove trunk-group 5** and press ENTER.

   The Trunk Group screen appears.
5. Press ENTER to remove the trunk group.

Inserting and absorbing digits

Use this procedure to modify the incoming digit string on DID and tie trunks by inserting (adding) or absorbing (deleting) digits. You'll need to do this if the number of digits you receive doesn't match your dial plan.

Instructions

As an example, let's say you have a DID trunk group. It's group number is 5. Your service provider can only send 4 digits, but your dial plan defines 5-digit extensions beginning with 6:

1. Type **change trunk-group 5** and press ENTER.

   The Trunk Group screen appears.

```
                                 TRUNK GROUP

Group Number: 5                Group Type: did              CDR Reports: y
  Group Name: Incoming calls          COR: 85        TN: 1        TAC: 105

                     Country: 1
                                Auth Code? n


TRUNK PARAMETERS
         Trunk Type: wink-start   Incoming Rotary Timeout(sec): 5
                                            Incoming Dial Type: tone
  Trunk Termination: rc              Disconnect Timing(msec): 500
    Digit Treatment: insertion                          Digits: 6
    Expected Digits: 4                          Sig Bit Inversion: none
Extended Loop Range? n      Trunk Gain: high    Drop Treatment: silence

 Disconnect Supervision - In? y
```

2. In the Digit Treatment field, type **insertion**.

   This field tells the switch to add digits to the incoming digit string. These digits are always added at the beginning of the string.

3. In the Digits field, type **6**.

   For insertion, this field defines the specific digits to insert. The switch will add a “6” to the front of the digit strings delivered with incoming calls. For example, if the central office delivers the string “4444,” Avaya MultiVantage will change it to “64444,” an extension that fits your dial plan.

4. In the Expected Digits field, type **4**.

   This field tells the switch how many digits the central office sends.

   NOTE:
   The Expected digits field doesn't appear on the screen for tie trunk groups.

5. Press ENTER to save your changes.

If your service provider sends 7 digits but you only need 5, you need to absorb the first 2 digits in the digit string. To absorb digits:

1. Type change trunk-group 5 and press ENTER.

   The Trunk Group screen appears.

2. In the Digit Treatment field, type **absorption**.

   This field tells the switch to remove digits from the incoming digit string. These digits are always removed from the beginning of the string.

3. In the Digits field, type **2**.

   For absorption, this field defines *how many* digits will be absorbed. The switch will remove the first 2 digits from the digit strings delivered with incoming calls. For example, if the central office delivers the string "556-4444," Avaya MultiVantage will change it to "64444," an extension that fits your dial plan.

4. In the Expected Digits field, type **7**.

   This field tells the switch how many digits the central office sends.

   NOTE:
   The Expected digits field doesn't appear on the screen for tie trunk groups.

5. Press ENTER to save your changes.

Related topics

Refer to ''Adding a DID trunk group'' on page 409 for instructions on administering a DID trunk group.

Refer to ''Adding a Tie or Access trunk group'' on page 415 for instructions on administering a tie trunk group.

Administering answer detection

Use this procedure to administer an outgoing or two-way trunk group for network answer supervision or answer supervision by timeout. If your network supplies answer supervision to a trunk group, you can administer Avaya MultiVantage to recognize and respond to that signal. If your network does not supply answer supervision, you can set a timer for all calls on that group. When the timer expires, the switch assumes the call has been answered and call detail recording starts (if you're using CDR).

For information about answer detection by call classification, contact your Avaya representative or refer to ''Answer detection'' on page 1352 for an introduction.

Before you start

Determine whether the trunk group receives answer supervision from your service provider or private network. For example, most loop-start CO, FX, and WATS trunks do not provide answer supervision.

Instructions

As an example, we'll administer trunk group 5 for both types of answer detection.

To administer trunk group 5 for answer supervision from the network:

1. In the Trunk Group screen for group 5, type **y** in the Receive Answer Supervision field.
2. Press ENTER to save your change.

Now let's administer answer supervision by timeout. We'll set the timer to 15 seconds. To administer trunk group 5 for answer supervision by timeout:

1. In the Trunk Group screen for group 5, type **15** in the Answer Supervision Timeout field.
2. Press ENTER to save your change.

Related topics

Refer to ''Answer detection'' on page 1352 for detailed information about this feature.

Administering trunks for listed directory numbers

Listed directory numbers (LDN) are the phone numbers given for an organization in public telephone directories. You can administer Avaya MultiVantage so that calls to different listed directory numbers go to the same attendant group. How you administer your system for LDN calls depends on whether the calls are coming in over DID and tie trunks or over CO and FX trunks.

Instructions

As an example, let's say that one attendant group answers calls for 3 different businesses, each with its own listed directory number:

- Company A — 855-2020
- Company B — 855-1000
- Company C — 855-1111

DID trunks and some tie trunks transmit part or all of the dialed digit string to the switch. If you want these calls to different numbers to go to one attendant group, you must identify those numbers for the switch on the Listed Directory Numbers screen.

Let's take the 3 businesses listed above as an example. Let's assume your switch receives 4 digits from the central office on a DID trunk group and that you're not using Tenant Partitioning. To make these calls to different listed directory numbers terminate to your attendant group:

1. Type **change listed-directory-numbers** and press ENTER.

   The Listed Directory Numbers screen appears.

```
                                                         Page    1 of    2
                          LISTED DIRECTORY NUMBERS

                            Night Destination:

       Ext         Name                           TN
    1: 2020        Company A                      1
    2: 1000        Company B                      1
    3: 1111        Company C                      1
    4:                                            1
    5:                                            1
    6:                                            1
    7:                                            1
    8:                                            1
    9:                                            1
   10:                                            1
```

2. In the Ext 1 field type **2020.**

   This is the LDN for Company A.

3. Type **Company A** in the Name field.

   This name will appear on the console display so the attendant knows which business the call is for and how to answer it.

4. Repeat steps 2 and 3 for the other two businesses.

   You can enter up to 20 different listed directory numbers on this screen.

5. Press ENTER to save your changes.

To make LDN calls over a CO or FX trunk group terminate to an attendant group, you must type attd in the Incoming Destination field on the Trunk Group screen for that group.

When you use the Listed Directory Number screen to assign some extensions to the attendant group, or when you enter **attd** in the Incoming Destination field on the Trunk Group screen for CO or FX trunks, Avaya MultiVantage treats these calls as LDN calls.

Related topics

Refer to ''Listed Directory Numbers'' on page 1627 for detailed information about this feature.

Managing announcements

13

Understanding announcements

An announcement is a recorded message a caller may hear while the call is in a queue, or if a call receives intercept treatment for some reason. An announcement is often used in conjunction with music.

Announcements can be integrated or external. Integrated announcements reside on a circuit pack in the switch carrier. External announcements are stored and played back from adjunct equipment. For more information on external announcements, see *Avaya MultiVantage™ Call Center Software Release 11 Guide to ACD Call Centers.*

Adding announcement data modules

NOTE:
This task is only required if you are using theTN750-series announcement circuit pack.

Your system uses a data module to save and restore the announcements from the integrated announcement circuit pack and your system's memory. You need to set up the data module that is built into the announcement circuit pack.

Before you start

You need to record your announcement on a special announcement circuit pack on your system, or on an external device like an answering machine. Refer to your *Avaya MultiVantage™ Hardware Solutions Guide* for circuit pack information.

Instructions

To set up the announcement data module, determine the port location of the Announcement circuit pack or the Auxiliary Trunk circuit pack. You can find circuit pack information on the Integrated Announcement Board screen.

To set up the data module on the Announcement circuit pack located at 01B18:

1. Type **add data-module next** and press RETURN**.**

   The Data modules screen appears.

   The system automatically fills in the number of the next available extension. Make sure the extension conforms to your dial plan. If you want to assign a specific extension, type that extension in the command instead of "next."

```
                                DATA MODULE

Data Extension: 2002                     Name: announcement data module
          Type: announcement               COS: 1
         Board: 01B18                      COR: 1
           ITC: restricted__                TN: 1



  ASSIGNED MEMBER (Station with a data extension button for this data module)

           Ext      Name
       1: ________ __________________________
```

2. In the Name field, type **announcement data module**.
3. In the Type field, type **announcement** and press RETURN.

   The Port field automatically changes to Board.
4. In the Board field, type **01B18**.

   This is the address of the announcement circuit pack.
5. Press ENTER to save your changes.

Adding announcement extensions

You need to assign an extension for each announcement that you want to record. After you define an announcement extension, you use it to record and access the announcement.

Announcements can be stored on a TN2501AP circuit pack, TN750 circuit pack or, for Avaya™ S8100 Media Server only, on the Integrated Scalable Speech Processor Application (ISSPA) circuit pack.

Instructions

Let's use extension 1234 for an announcement about business hours. We'll use the integrated announcement circuit pack located on 01B18 (for Avaya™ S8100 Media Server, we might use 1A13).

To add an announcement extension 1234:

1. Type **change announcements** and press RETURN.

   The Announcements/Audio Sources screen appears.

```
                      ANNOUNCEMENTS/AUDIO SOURCES

   Ext.       Type       COR  TN   Name              Q  QLen Pro Rate Port
1: 1234____   integrated 1_   1_   business hours    n  N/A  n   32   01B18
2: ________   _________  1_   1_   _______________   n
3: ________   _________  1_   1_   _______________   n
4: ________   _________  1_   1_   _______________   n
5: ________   _________  1_   1_   _______________   n
```

2. In the Ext field, type **1234**.
3. In the Type field, type **integrated.**
4. In the Name field, type **business hours.**

   NOTE:
   If you are using either the TN2501AP or Avaya™ S8100 Media Server ISSPA circuit packs, you must enter a name in the Name field. This name becomes the filename for the announcement file. See ''Announcements/Audio Sources'' on page 588 for more information on requirements for this field.

5. In the Q field, type **y.**

   This sets up an announcement queue so calls wait in a queue to hear an announcement if all the ports on the announcement circuit pack are busy.

   If you leave this field n, there is no announcement queue. When all the announcement ports are busy the caller hears busy or other feedback depending upon how the announcement was accessed.

6. N/A appears in the QLen (queue length) field. You cannot change this field for integrated announcements because they have a pre-set queue length.

7. In the Pro (protected) field, type **n**.

   If you enter **n**, users with console permissions can change the announcement. If you enter **y**, the announcement cannot be changed.

   This field appears only if the Type field is integrated.

8. In the Port field, type:

   - the location of the announcement circuit pack (TN2501AP or TN750). In this example, we use **01B18**.
   - **1A13** if you are using the Avaya™ S8100 Media Server ISSPA circuit pack

9. Repeat steps 2 -8 with the correct information for each announcement you want to record.

10. Press ENTER to save your work.

NOTE:
These steps have only created the administered name for the announcement file. You fill the file space when you record an announcement or transfer an announcement file to the circuit pack through an FTP session.

To check that the announcement administration is correct:

1. At the SAT, type **list integrated-annc-boards** and press RETURN.

Recording announcements

You can record an announcement for callers to hear when they dial a specific extension. You can use the same steps to change an existing announcement.

For instructions on how to record an announcement at a computer or from a phone, to a TN2501AP, see ‘‘Recording VAL announcements’’.

If you want to change an announcement that is recorded on a circuit pack without memory, you have to delete the original message and record a new announcement. Refer to ‘‘Deleting and erasing announcements’’ on page 440.

Announcement sessions always use port 0, which also is used for playing announcements, on the TN750 circuit packs. Announcement sessions always use port 1, which is dedicated for telephone access, on the TN2501AP circuit packs. With the TN2501AP circuit pack, the port is only busy if another telephone access session is active.

To begin an announcement session, the user must dial the administered feature access code (FAC) followed by the announcement session. If an announcement session is already in progress, or if a save or restore command is in progress, then the user hears reorder tone (fast busy) and the system drops the call.

If the telephone session port to an integrated circuit pack is in use, then the user hears reorder tone followed by silence. This indicates that the port will be reserved for an announcement session. The user should redial the FAC and extension every 45 seconds to gain access to the port.

NOTE:
For the May and later release of MultiVantage™ software, multiple telephone sessions are allowed, with one session associated with each active integrated announcement circuit pack.

Once an end user accesses an announcement session, the user can dial **1** to record an announcement, **2** to play an announcement, or **3** to delete the announcement. If the circuit pack memory is more than 90% full, then the switch gives stutter dial tone when the user gains access to an announcement session. Even if the user hears stutter tone, the user should begin speaking to record the announcement.

Before you start

- You need to have a phone or console with a class of service (COS) that provides console permissions to record announcements.

  Refer to ‘‘Command Permission Categories’’ on page 655 for more information on COS permissions.

Instructions

To record an announcement to extension 1234, or change the announcement already recorded there, we’ll use a phone with console permissions. In our example, the announcement feature access code is *56.

To record or change the announcement:

1. Dial ***56** from a phone or console.
   - If you hear dial tone, go to step 2.
   - If you hear a fast busy signal, hang up and redial the FAC and extension every 45 seconds until you hear dial tone.

2. Dial the announcement extension **1234**.

   You hear dial tone.

3. Dial **1** to begin recording.

   - If you hear a beep or stutter tone, begin speaking. If the circuit pack memory becomes full during recording, the system drops your connection and does not retain the announcement.
   - If you hear intercept tone, hang up and record your announcement on another extension.

4. To end the recording:

   - If you are using a digital phone, press **#**. You hear dial tone.
   - If you are using an analog phone, hang up. If your analog phone is not connected through lineside DS1, the system records an electrical click at the end of the recording.

5. To listen to the announcement you just recorded:

   - If you are using a digital phone, do not hang up. Press 2. The recording plays back through the handset.

6. If you are not satisfied with the announcement,

   - press **1** to re-record the announcement.
   - press **3** to delete the announcement and end the recording session.

7. If you want to listen to the announcement after you have hung up, dial the extension from any phone or console. In this example, dial **1234**. The announcement plays through the handset.

   NOTE:
   You have to wait 15 seconds after you record the announcement before you can dial the extension to hear your announcement. During this 15-second window, you cannot record a new announcement and no one can play this announcement. You can re-record the announcement. Dial the feature access code, dial the extension, and press **2** before the 15-second timer expires.

Saving announcements

NOTE:
This task is only applies if you are using a TN750-series announcement circuit pack.

If you are using the TN2501AP (VAL) announcement circuit pack, see ''Moving announcements from the VAL circuit pack'' on page 451.

If you are using the ISSPA announcements capability of the Avaya™ S8100 Media Server, see ''Backup via the Web interface (Avaya™ S8100 Media Server only)'' on page 53.

You can save and back-up announcements from an announcement circuit pack to system memory. Your system memory is a tape, disk, or memory card, depending on your system. Use this procedure primarily for announcement circuit packs without built-in memory.

CAUTION:
Announcements that are recorded on announcement circuit packs without built-in memory are lost if they are not saved to system memory before power is shut down or the circuit pack is reset or removed.

For extra security, you may also want to save announcements from announcement circuit packs that have their own built-in memory to another announcement circuit pack with built-in memory or to tape.

CAUTION:
Do not copy, save, or restore announcements from an announcement circuit pack that has built-in memory to one without built-in memory because it may corrupt the announcement data.

Instructions

To save or backup announcements when you have only one announcement circuit pack:

1. Type **save announcements** and press RETURN to save the changes.

   This process can take up to 40 minutes. You cannot administer your system while it is saving announcements.

To save or backup announcements when you have more than one announcement circuit pack, specify the circuit pack address in your command.

For example to save the announcements from the circuit pack at 01B18:

1. Type **save announcements from 01B18** and press RETURN.

   NOTE:
   If you have announcement circuit packs with and without built-in memory, save the circuit pack without built-in memory.

If you have a duplicated system with an active processor and a stand-by processor, you can save announcements from the announcement circuit pack to both processors. Simply add system memory for both processors to the save announcements command.

If the save announcements procedure fails on one of the processors, it continues on the other processor, which results in inconsistent announcement data between the two processors.

To verify that the save announcements command was successful:

1. Type **display integrated announcement boards** and press RETURN.

   The Integrated Announcement Boards screen appears.

This screen shows the date, time, and location of the announcement circuit pack most recently saved. If the save announcement command failed or the save translations were not run before rebooting, “N/A” appears in these fields.

Copying announcements

NOTE:
This task only applies if you are using a TN750-series announcement circuit pack.

If you are using the TN2501AP (VAL) announcement circuit pack, see ‘‘Moving announcements from the VAL circuit pack’’ on page 451.

If you are using the ISSPA announcements capability of the Avaya™ S8100 Media Server, see ‘‘Backup via the Web interface (Avaya™ S8100 Media Server only)’’ on page 53.

You can copy recorded announcements between back-up disks and tapes when you want to store your recorded announcements in more than one place.

Instructions

Let's copy the announcements stored in our weekly disk back-up to the tape back-up we take off company premises for a month. To copy announcements between your back-up disk and tape:

1. Type **copy announcements** and press RETURN to save the changes.

   This process can take up to 40 minutes. You cannot administer your system while it is copying announcements.

Restoring announcements

NOTE:
This task only applies if you are using a TN750-series announcement circuit pack.

If you are using the TN2501AP (VAL) announcement circuit pack, see ''Moving announcements to a VAL circuit pack or to another LAN device'' on page 453.

If you are using the ISSPA announcements capability of the Avaya™ S8100 Media Server, see *''Backup via the Web interface (Avaya™ S8100 Media Server only)'' on page 53*.

You can restore announcements from system memory to an announcement circuit pack. Your system memory is a tape, disk, or memory card, depending on your system.

If you have a duplicated system, you system always restores the announcements located on the active processor.

Instructions

Let's restore announcements from our system memory (stored on a disk) to the integrated announcement circuit pack on our system.

To restore announcements from system memory to the integrated announcement circuit pack:

1. Type **restore announcements disk** and press RETURN.

Let's restore announcements from system memory to an announcement circuit pack that has built-in memory. We know that the announcement circuit pack is located at 01B18:

1. Type **restore announcements from cabinet 01. carrier B, slot 18** and press RETURN.
2. Press ENTER to restore announcements.

Fixing problems

Problem	Possible causes and solutions
You receive error code E28	You do not have an integrated announcement circuit pack in the system.
You receive error code E31	A call is connected to the announcement on the circuit pack and the port is busy. Wait for the call to disconnect and try restore announcements again.

If the system crashes or there is a processor interchange during restore announcements, the restore fails and there is no valid announcement on the circuit pack. Repeat the process when the system is operating properly.

Deleting and erasing announcements

You can use a system phone to delete recorded announcements from an announcement circuit pack that does not have stored memory. When you delete the announcement from the circuit pack, you do not delete the announcement from your system backup tape or disk.

You can also erase the announcements stored on an integrated announcement circuit pack that has stored memory.

NOTE:
The system denies any attempt to delete an announcement while it is playing, being transferred, or backed up to FLASH (amber LED flashes), regardless of whether the attempt is from a system phone, the SAT, or through an FTP session.

Before you start

- Look up the announcement extension, which is mapped to a specific TN2501AP circuit pack by location.
  1. At the SAT, type **list integrated-annc-boards** and press RETURN.
  2. Determine the extension(s) for the announcement(s) that you want to delete.
- Record the feature access code (FAC) for an announcement session.

Instructions

Let's use a phone with console permissions to delete the recorded announcement assigned to extension 1234. In our example, we know that the announcement feature access code is *56.

1. Dial ***56** from a phone or console.

   You hear dial tone.
2. Dial **1234**.

   You hear dial tone.
3. Dial **3** to delete the announcement from the announcement circuit pack.

   You hear a confirmation tone.

   If the announcement is protected or is playing at the time of the command, you hear a fast busy signal (reorder tone) and the system does not delete the announcement.
4. Hang up the phone.
5. To ensure that an announcement was deleted, dial **1234**.

   You hear a busy signal if the announcement was deleted.
6. Repeat Steps 1-5 for each announcement that you want to delete.

   You can delete only one announcement at a time.

You may also want to remove the announcement extension from the system. To remove the extension, use your system administration terminal to complete the following steps:

1. Type **change announcements** and press RETURN.

   The Announcements/Audio Sources screen appears.
2. Delete the information in the Ext and Type fields.
3. Press ENTER to save your work.

Finally, to erase announcements on the announcement circuit pack located at 01B18:

1. Type **erase announcements 01B18**.

   A warning message appears.

2. Press ENTER to erase the announcements on the circuit pack.

Setting up continuous-play announcements

You can set up announcements to keep on repeating while callers are connected to the announcement, so a caller listens until the system plays the entire announcement.With a “barge-in” queue, you do not need a separate port for each announcement.

For example, you can set up an Automatic Wakeup announcement to repeat and use a barge-in queue. When guests pick up the phone to hear an announcement at a particular time, they use only one port and the message repeats on that port until the last guest goes off-hook and the message ends.

Before you start

You must use an integrated, multiple-integrated, or external announcement for barge-in announcements.

Instructions

Let’s set up a continuous-play announcement for our integrated announcement on extension 1234:

1. Type **change announcements** and press RETURN.

   The Announcements/Audio Sources screen appears.

```
                          ANNOUNCEMENTS/AUDIO SOURCES
    Ext    .    Type       COR  TN   Name             Q  QLen Pro Rate Port
 1: 1234____    integ-rep   1_  1_   business hours   b  N/A  n   32   01B18
 2: ________    _________   1_  1_   ______________ n
 3: ________    _________   1_  1_   ______________ n
 4: ________    _________   1_  1_   ______________ n
 5: ________    _________   1_  1_   ______________ n
 6
```

2. Type **b** in the Q field on the same line as extension 1234.

3. Leave **business hours** in the name field, or enter a new description for the announcement.

4. Press ENTER to save your work.

Getting started with the TN2501AP

Before you can use the capabilities of the TN2501AP announcement circuit pack, it must be properly installed and configured. These instructions are contained in other documents in the Avaya MultiVantage documentation library. For more information about these and other tasks related to using the TN2501AP, refer to the documents listed in the following table.

Task	Information source
Installing the TN2501AP Administering IP Connections Adding IP Routes Testing the IP Connections	*Made Easy Tool for DEFINITY® Media Server Configurations* *Installation, Upgrades and Additions for Avaya™ CMC1 Media Gateways*
Administering Announcements (recording, copying, deleting, etc.)	“Managing VAL Announcements Using the SAT” on page 443 and “Managing VAL Announcements Using FTP” on page 449
Viewing announcement usage measurements (**list measurements announcement** command)	*Reports for Avaya MultiVantage™ Software*
Troubleshooting announcements	“Troubleshooting VAL Announcements” on page 457
Troubleshooting VAL hardware	*Maintenance for Avaya™ MultiVantage and DEFINITY® Server* for your switch model

Managing VAL Announcements Using the SAT

These are the tasks that you can perform from a System Access Terminal (SAT):

- Administering an announcement using the SAT
- Recording announcements (with an option to use a system phone)
- Deleting announcements (with an option to use a system phone)

To administer an announcement using the SAT, see ‘‘Adding announcement extensions’’.

Recording VAL announcements

You can record an announcement for callers to hear when they dial a specific extension or as part of call vectoring. You can use the same steps to change an existing announcement.

You can record announcements in many ways:

- Professional or computer recordings
- Recording new announcements at a computer
- Recording announcements at a system phone

Before you start

Ensure that the announcement administration is complete before proceeding. You must have assigned a name before you can record an announcement (see ‘‘Adding announcement extensions’’ on page 433).

If you are replacing a TN750C announcement circuit pack with the TN2501AP,

- get a list and description of the announcements stored on the TN750C circuit pack.
- re-record the announcements on a computer or at a professional recording studio as .wav files (CCITT µ-Law or A-Law, 8KHz, 8-bit mono), so that they are ready to transfer to the new announcement circuit pack after it is installed and administered.

Replacing old announcement circuit packs with the new TN2501AP circuit pack requires that you

- remove previous announcement administration
- record new announcements for the TN2501AP
- re-record any announcements currently resident on the TN750 circuit packs that you are replacing. You cannot transfer or restore TN750 announcements from flash card, tape, or optical disk to the TN2501AP.

Announcement file format requirements

In order to be compatible with the TN2501AP circuit pack and the MultiVantage system, announcement recordings must have the following parameters:

- CCITT A-Law or CCITT μ-Law companding format (do not use PCM)
- 8KHz sample rate
- 8-bit resolution (bits per sample)
- Mono (channels = 1)
- The μ-Law (Mu-Law) companding is used in the United States and A-Law is used internationally. Use the companding format specified on the System Parameters Country Options screen.

Announcements that are recorded in this format occupy 8Kbps of file space. For example, a 10-second announcement creates an 8K .wav file.

Professional or computer recordings

In order to be compatible with the TN2501AP circuit pack and the MultiVantage system, announcement recordings must meet the ‘‘Announcement file format requirements’’ described above.

Recording new VAL announcements at a computer

To record an announcement at a computer:

1. At the computer, open the application that you use to record .wav files.
2. Set the recording parameters.
3. Record the announcement by speaking into a microphone connected to the computer.
4. Play the announcement back at the computer before transferring the file to the VAL circuit pack.

Recording VAL announcements at a system phone

To record an announcement at a system phone, see ‘‘Recording announcements’’.

Converting announcement files to VAL format

If you are sharing recordings in a multi-site environment with Avaya MultiVantage and Avaya™ S8100 Media Server or CONVERSANT systems, you must convert announcement files for use on either system. If you need to convert an announcement file to the required format, you can use a sound recording utility application to do so.

To convert a previously-recorded or an Avaya™ S8100 Media Server- or CONVERSANT- compatible announcement file to Avaya MultiVantage-compatible formats:

1. Open the sound recording application on your computer (for example, Microsoft Windows Sound Recorder).
2. Open the file you want to convert.
3. Check the file properties to see if you need to change the parameters.
4. If you need to change the recording parameters, look for a conversion tool (some have a Convert Now option, others use Save As).
5. Change the file parameters to those listed in ‘‘Announcement file format requirements’’.

   NOTE:
   In some applications, assigning the format (for example, CCITT μ-Law) sets the remainder of the default parameters. Check each parameter carefully, perhaps changing default settings to match the parameters listed above. CCITT μ-Law or A-Law can be referred to as ITU G.711 μ-Law or ITU G.711 A-Law, respectively.

Converting announcements for Avaya™ S8100 Media Server/CONVERSANT

Avaya™ S8100 Media Server and CONVERSANT have a recording conversion utility that supports file formats similar to those required by VAL. However, the conversion utility can only read PCM-format announcement files.

If you are converting an announcement file for use on Avaya™ S8100 Media Server or CONVERSANT systems:

1. If the file’s companding format is already PCM, go to Step 5.

   If you are not sure what the file format is, proceed with Step 2.
2. At a computer, open the sound recording application (for example, Microsoft Windows Sound Recorder).
3. Open the file that you want to convert.

4. Save (Convert Now or Save As) the announcement with these formats:
   - Format: PCM
   - Bits/Sample: 8
   - Sample Rate: 8KHz
   - Mono (channels = 1)

   NOTE:
   The recording conversion utility requires that announcement files are in PCM format.

5. Open the file in recording conversion utility.
6. Convert the file to SSP format.

Deleting VAL announcements

Before you start

Look up the announcement information:

1. At the SAT, type **list directory board** and press RETURN.
2. Determine which announcement(s) that you want to delete, either by extension or filename.
3. Decide whether you are
   - Deleting individual announcement files using the SAT.
   - Deleting all announcements on a circuit pack using the SAT.
   - “Deleting and erasing announcements”

NOTE:
The system denies any attempt to delete an announcement while it is playing, being transferred, or backed up to FLASH (amber LED flashes), regardless of whether the attempt is from a system phone, the SAT, or through an FTP session.

Deleting individual VAL announcement files using the SAT

To delete the announcement named Closed (announcement #3 on Screen 13):

1. At the SAT, type **remove file board *board-location* /annc/*filename.wav*** and press RETURN.

   For example, to delete announcement #3 in Screen 13, type:

   remove file board 01A11 /annc/Closed.wav

   NOTE:
   Filenames are case-sensitive and require the .wav file extension.

   The */annc* portion of the command directs the system to the announcement subdirectory on the VAL circuit pack, and */Closed.wav* indicates to delete the filename Closed.wav.

Deleting all VAL announcements on a circuit pack using the SAT

To delete all of the announcement files on the VAL circuit pack:

1. At the SAT, type **busyout board *board-location*** and press RETURN.

   Ensure that the command is successful.

   NOTE:
   When the VAL board is busied out,

   - both the RSCL and ethernet ports are busied out.
   - firmware takes down the ethernet link.
   - FTP is disabled because the ethernet link is down.
   - announcements on that circuit pack cannot play.

2. At the SAT, type **erase announcements board *board-location*** and press RETURN.

   CAUTION:
   This command deletes the specified announcement file in both RAM and FLASH memory. The board firmware ignores the protect flag (Pro field) when erasing the announcement files.

3. At the SAT, type **list directory board** and press RETURN.
4. Verify that there are no files listed.

   NOTE:
   The announcement directory on the TN2501AP is **/annc**.

5. Type **list integrated-annc-boards** and press RETURN.

   Check the list to see that the announcement was deleted. The Length in Seconds field should show 0.

Deleting VAL announcements at a system phone

See ‘‘Deleting and erasing announcements’’.

Managing VAL Announcements Using FTP

This section includes information on setting up and terminating a file transfer protocol (FTP) session and outlines tasks that you can do in an FTP session.

There are 3 basic components to an FTP session:

- Setting up an FTP session
- Performing tasks in an FTP session
- Ending an FTP session

SECURITY ALERT:
Be sure to read and observe all of the Security Alerts regarding enabling and disabling the TN2501AP filesystem and FTP sessions into it.

Setting up an FTP session

Setting up a file transfer protocol (FTP) session into the VAL circuit pack involves:

1. Preparing the VAL circuit pack for the FTP session, which
   - allows an FTP session on an individual VAL circuit pack.
   - creates an ftp-login and ftp-password for that session.
2. Starting an FTP session from a computer or network management terminal. Before you can start the FTP session, you need to know
   - the VAL circuit pack’s IP address from Step 1.
   - the VAL circuit pack’s ftp-login and ftp-password from Step 1.

Preparing the VAL circuit pack for the FTP session

To prepare the VAL circuit pack for the FTP session, including setting the username and password:

1. At the SAT, type **enable filesystem board *board-location* login *ftp-username [3-6 characters] ftp-password [4-11 characters]*** and press RETURN.

   For example, the command:

   enable filesystem board *01A11* login *romeo shakespeare*

   enables an FTP session into the VAL circuit pack in Cabinet 1, carrier A, slot 11. The ***ftp-username*** (3-6 characters) for this session is romeo, and the ***ftp-password*** (4-11 characters) is shakespeare.

   When the FTP session on the circuit pack is enabled, the announcement and firmware files are available to anyone who knows the VAL circuit pack's IP address, the ftp-username, and the ftp-password.

SECURITY ALERT:

We recommend using a unique ftp-login and ftp-password for each FTP session.

Starting an FTP session

If you are unfamiliar with FTP client application software, contact your network administrator for information about access to an FTP session.

The following points apply to FTP sessions into the VAL circuit pack:

- In FTP sessions, filenames are case-sensitive and require the .wav" file extension.
- Only one FTP session can be active at a time. If an FTP session is already active for a particular VAL circuit pack, the system denies a second attempt to establish an FTP session from some other remote host.
- The VAL circuit pack has two user-accessible directories:
  - — /annc for playable announcements
  - — / (root) for temporary storage of embedded software updates. Use this directory only for software updates.
- FTP sessions time out after 10 minutes of inactivity.

Instructions

1. At the FTP client, type **ftp *val-ip-address*** and press ENTER. The IP address must match the switch-administered IP address (see **change node-names ip**).
2. At the username prompt, type **romeo** and press RETURN.
3. At the password prompt, type **shakespeare** and press RETURN.

   The system responds with User logged in.

   NOTE:
   Once you are logged in you are in the announcements directory (/annc).

4. If you are moving files to or from the VAL circuit pack, you must set the system to binary mode. At the FTP client, type **bin** and press RETURN.

   The system responds with Types set to I, binary mode.

   CAUTION:
   If you do not transfer announcement files in binary mode, they can be corrupted and the FTP session can fail.

Performing tasks in an FTP session

You are now ready to perform any of these tasks in the FTP session

- Moving announcements from the VAL circuit pack
- Deleting announcements
- Moving announcements to a VAL circuit pack or to another LAN device
- Combining tasks

Moving announcements from the VAL circuit pack

When you move a file from the VAL circuit pack, you are either

- backing up (archiving) an announcement file.
- copying an announcement to another VAL circuit pack (restoring).

Moving a file in an FTP session means copying the file from the VAL circuit pack to the FTP client's default directory. If you want to move the file to another circuit pack or LAN device, see ''Moving announcements to a VAL circuit pack or to another LAN device''.

Before you start

- Ensure that the steps in ''Setting up an FTP session'' are complete.
- Ensure that you have adequate storage space for backup on the remote host computer. Depending on the number of announcements and file type, 60 minutes of recording may require up to 32 megabytes of storage space for backup on the remote host computer.
- Know the IP address and location of the TN2501AP circuit pack as well as the filename (**list directory board**) for the announcement that you want to move.

  NOTE:
  The announcement directory on the TN2501AP circuit pack is ***/annc***.

Instructions

To backup or save an announcement from the VAL board to the client computer through an FTP session:

1. Ensure that the steps in ''Setting up an FTP session'' are complete.
2. At the FTP client, type **get *filename.wav*** and press RETURN.

   Example: **get Closed.wav**

   The announcement file is written to the directory from which you initiated the FTP session.

   NOTE:
   FTP upload or download of announcement files does not preserve the created timestamp. The file receives the current date and time when it is written to the circuit pack or on the computer.

3. List the FTP client directory contents and ensure that the announcement file is among those listed.
4. Terminate the FTP session (see ''Ending an FTP session'').

Deleting VAL announcements using FTP

You can delete an announcement from a TN2501AP circuit pack or from a LAN device.

NOTE:
The system denies any attempt to delete an announcement while it is playing, being transferred, or backed up to FLASH (amber LED flashes), regardless of whether the attempt is from a system phone, the SAT, or through an FTP session.

Before you start

- Know the IP address, the announcement filename that you are deleting, and the VAL circuit pack location (**list directory board**).

Instructions

To delete an announcement on a TN2501AP circuit pack through an FTP session:

1. Ensure that the steps in ''Setting up an FTP session'' are complete.
2. At the computer client, type **delete *filename.wav*** and press RETURN.

   Example: **delete Closed.wav**

   NOTE:
   The announcement file is only removed from volatile RAM memory. Approximately 5 minutes later, the file is removed from nonvolatile ROM flash memory.

3. List the contents of the announcement directory and ensure that the file is not listed.

   NOTE:
   The .wav file extension on the announcement files are visible when you view the announcement directory from the FTP client.

4. Terminate the FTP session (see ''Ending an FTP session'').
5. At the SAT, type **change announcements** and press RETURN.

   The Announcements/Audio Sources screen appears.

6. Remove the announcement administration by deleting the entire line associated with the announcement.
7. Press ENTER to save your changes.

Moving announcements to a VAL circuit pack or to another LAN device

You can copy an announcement file to the VAL circuit pack to another LAN device.

Before you start

- Know the announcement filename and its location on the client computer.
- Know the destination IP address, the filename, and circuit pack location of the announcement and VAL circuit pack to which you are moving the announcement (**list directory board**).

- Ensure that you *have not just* administered the announcement on the Announcements/Audio Sources screen. If announcement administration precedes the file transfer, then
  - — the announcement appears with a zero (0) length on the **list integrated-annc-boards** screen.
  - — The Time Remaining fields on both the **list integrated-annc-boards** and **display integrated-annc-boards** screens do not refresh to reflect the presence of the new announcement file on the circuit pack.

  Use this procedure to ensure that the announcement length is accurate:

  1. Administer the announcement at the switch (**change announcements**), using the identical filename in the Name field *without spaces or the .wav file extension*.
  2. Attempt to play the announcement that was administered first and transferred second.

     The switch returns a busy signal at the first play attempt.
  3. Attempt to play the announcement that was administered first and transferred second in a telephone access session.

     The switch returns a busy signal at the first play attempt.
  4. Re-record this announcement with the same filename at a phone (see ‘‘Recording VAL announcements at a system phone’’).

Instructions

To copy an announcement to a VAL circuit pack or to another LAN device:

1. Ensure that the steps in ‘‘Setting up an FTP session’’ are complete.
2. At the FTP client, type **put *filename.wav*** and press RETURN.

   Example: **put Closed.wav**
3. List the contents of the VAL announcement directory or LAN device and look for the announcement file among those listed.

   NOTE:
   FTP upload or download of announcement files does not preserve its timestamp. The file receives the current date and time when it is written to the circuit pack or to a computer.
4. After you are sure that the announcement is on the VAL circuit pack, administer the announcement at the switch (**change announcements**), using the identical filename in the Name field *without spaces or the .wav file extension*.
5. Terminate the FTP session (see ‘‘Ending an FTP session’’).

Combining tasks

When you combine copying (the **get** command) and moving (the **put** command) announcement files, you can rearrange VAL announcements.

Before you start

- Know the IP address, the filename, and location of the destination VAL circuit pack to which you are moving an announcement (**list directory board**).

Instructions

To move an announcement to a VAL circuit pack from another VAL circuit pack in an FTP session:

1. Ensure that the steps in ‘‘Setting up an FTP session’’ are complete.

   NOTE:
   You must first establish an FTP session into the circuit pack *from which you are restoring an announcement*.

2. List the directory contents and ensure that the announcement file is among those listed.
3. At the FTP client, type **get *filename.wav*** and press RETURN.

   Example: **get Closed.wav**

   A copy of the file is written to the directory from which you initiated the FTP session.
4. List the FTP client directory contents to ensure that the announcement is among those listed.
5. Terminate the FTP session (see ‘‘Ending an FTP session’’) to the circuit pack *from which you copied the announcement file*.
6. Set up a new FTP session into the destination VAL circuit pack (see ‘‘Setting up an FTP session’’).
7. At the FTP client, type **put *filename.wav*** and press RETURN.

   Example: **put Closed.wav**
8. List the VAL announcement directory contents to ensure that the announcement is among those listed.
9. Terminate the FTP session to the circuit pack *to which you copied the announcement file*.

Ending an FTP session

FTP sessions to a VAL circuit pack originate at the FTP client end. You terminate an FTP session by

- logging out from the FTP client (type **bye** or **quit** and press ENTER)

  and

- typing **disable filesystem board *board-location*** at the SAT and press RETURN. (This clears the ftp-username and ftp-password.)

  or

- you can effectively terminate the session from the Avaya MultiVantage end by letting the system time out (10 minutes of inactivity).

SECURITY ALERT:
Both logging out of the FTP session and disabling the VAL circuit pack filesystem provide a higher degree of security.

NOTE:
If you only disable the circuit pack filesystem, you can continue your FTP session. However, new FTP session logins are not allowed.

VAL Manager

VAL Manager is a standalone application that allows you to copy announcement files and MultiVantage announcement information to and from a MultiVantage system over a LAN connection.

VAL Manager offers the following basic features:

- Simplified administration to add, change, and remove MultiVantage announcement information.
- The ability to back up and restore announcement files and information to and from a MultiVantage system.
- The ability to view the status of announcement on the VAL circuit pack in any MultiVantage system.

See your Avaya representative for more information about VAL Manager.

Troubleshooting VAL Announcements

If a working announcement file is deleted via FTP, the next attempt to play the announcement fails, and the system adds a software event to the Denial Events Log. You can view this log to see if the announcement has been deleted, and to see if other events have occurred related to announcements.

Viewing the Denial Events Log

To view the Denial Events Log:

1. At the SAT, type **display events** and press RETURN.

   The Events Report screen appears. This input screen helps you focus the report on events of a certain type or from a certain time period.

```
display events                                  Page   1 of   1    SPE B

                         EVENT REPORT

  The following options control which events will be displayed.

         EVENT CATEGORY

            Category: denial

         REPORT PERIOD

            Interval: ___     From: __/__/__:__  To: __/__/__:__


         SEARCH OPTIONS

                           Vector Number: ___
                              Event Type: _____
```

Screen 26. Event Report screen (display events)

2. In the Category field, select or type **denial**.
3. You can further limit the report by setting the Interval field to one of the following selections (select from the help list or type the first letter):
   - **all**
   - **month**
   - **day**
   - **hour**
   - **minute**

4. Press ENTER.

   The Events Report screen appears.

```
display events                                               Page   1  of  1

                                 EVENTS REPORT

Event Event                      Event     Event      First       Last     Evnt
Type  Description                Data 1    Data 2     Occur       Occur     Cnt

2028  Annc file has bad format  8C0009    0        05/14/11:55 05/14/12:04  2
2027  Annc not found on board   8C0007    0        05/14/12:09 05/14/12:09  1
```

Screen 27. Events report screen (display events)

5. Look at the 2027 entry in the Event Type field.
   - The Event Description field explains that the announcement is not on the circuit pack.
   - The Event Data 1 field contains the announcement number (hexadecimal in the lower three digits).

NOTE:
This denial event only appears once in the Denial Events Log.

Troubleshooting poor sound quality

If you played an announcement files back in another environment and it sounded great, but when you play it back in the MultiVantage environment the sound quality is poor, ensure that the file formats are compatible. A good announcement file format must be:

- 8Kbps sample rate
- 8-bit resolution (bits per sample)
- A-law or Mu-law companding format
- Mono (channels = 1)

You must also have the same companding mode administered on page 1 of the System-Parameters Country-Options screen (**change system-parameters country-options**).

The system records a software denial event in the Denial Events Log each time it plays an announcement with a bad file format. Refer back to the Events report screen (display events) and find the 2028 entry in the Event Type field:

- The Event Description field explains that the announcement has a bad file format.
- The Event Data 1 field contains the announcement number (hexadecimal in the lower three digits).

Viewing announcement measurements

You can view a report of announcement measurements, including how many times an announcement was queued to play, how many callers dropped while in queue, and how many times all announcement ports were busy during the report period.

For more information about this report, see *Reports for Avaya MultiVantage Software* on your documentation CD.

Related topics

For more information, see ‘‘Announcements’’ on page 1348.

Non-volatile storage of announcements on VAL

The VAL board has a RAM based file system on which the announcements are stored. This is where a newly recorded or FTP’d announcement resides. The VAL constantly checks to determine whether changes have been made to the file system (file added or deleted or renamed). Once a change has been detected, the VAL waits for 5 minutes of stability. After such a stable period, the VAL backs up the RAM based file system to its FLASH. The FLASH version of the file system is non-volatile and is what is restored upon reset/reseat.

If you record a new announcement and immediately reset the board, the announcement will be lost. The yellow LED blinks while a backup from RAM to FLASH occurs. All of the above is true regardless which (if any) of the .wav files on the board are administered announcements. It is just the way the board (firmware) operates. Note that only files with a .wav extension in the /annc directly are backed up to FLASH.

MultiVantage announcement synchronicity

As mentioned above, there is a separation between the actual files on the VAL board and the set of administered announcements. You can have 100 .wav files on a VAL and only be using (administered) 5 of them. The best way to visualize this is to look at the difference between a **list integrated** and the **list directory board <brdloc>/annc** commands.

By going into the Change Announcement screen and changing the name of an announcement, what is happening? You are changing only the administration of the announcement. This means when the extension is called, the new file name is requested. The board still has a file with the old file name but not one with the new file name. You then could run **change announcement** again and switch the name back to the old one and the announcement will play fine. The VAL board is completely unaware that you change the **change announcement** command.

Changing the name of the announcement

To change the name of an announcement you must do two things.

1. Change the administered name (using the **change announcement** command)
2. Change the name of the file on the VAL board

Changing the name of the file on the VAL board

There are two ways to accomplish this.

> **NOTE:**
> For the following steps, do not use the FTP **rename** command since the FTP server on the VAL board does not support it.

1. Change the name of the existing file using the PC.
   a. Access the file on the PC if already there or FTP the announcement file from the VAL board (in binary mode) to the PC.
   b. Rename it locally on the PC.
   c. FTP the file with the new name back to the VAL board.
   d. Assign the new name to the existing extension on the announcement screen
   e. Delete the old file on the board using the **remove file** command, if you wish, or

2. Re-record the announcement with the new file name.
   a. FTP the new announcement to the VAL board.
   b. Assign the new name to the existing extension on the announcement screen (replacing the old file name).
   c. Delete the old announcement file using the **remove file** command (if desired).

Managing group communication

14

Group communication features allow co-workers to communicate with each other more efficiently. This chapter shows you how to administer your system so users can place a variety of pages from their phones or use their phones as intercoms. You can also give specific users permission to monitor other users’ calls or to interrupt active calls with important messages.

Setting up voice paging over loudspeakers

Use this procedure to allow users to make voice pages over an external loudspeaker system connected to your Avaya MultiVantage system. If you’re using an external paging system instead of an auxiliary trunk circuit pack, don’t use this procedure. External systems typically connect to a trunk or station port and are not administered through the Loudspeaker Paging screen.

Before you start

Your switch must have one or more auxiliary trunk circuit packs with enough available ports to support the number of paging zones you define. Each paging zone requires 1 port. Refer to the *Avaya MultiVantage™ Hardware Solutions Guide* for information on specific circuit packs.

Instructions

As an example, let's set up voice paging for an office with 5 zones. We'll allow users to page all 5 zones at once, and we'll assign a class of restriction of 1 to all zones.

1. Type **change paging loudspeaker** and press RETURN.

   The Loudspeaker Paging screen appears.

```
                          LOUDSPEAKER PAGING

                                CDR? y
          Voice Paging Timeout (sec): 30
         Code Calling Playing Cycles:

PAGING PORT ASSIGNMENTS
                 Voice Paging     Code Calling
 Zone  Port      TAC  COR  TN     TAC  COR  TN     Location:
   1:  01C0501   301   1   1                1      Reception area
   2:  01C0502   302   1   1                1      Shipping & receiving
   3:  01C0503   303   1   1                1      Staff offices
   4:  01C0504   304   1   1                1      Management suite
   5:  01C0601   305   1   1                1      Breakroom
   6:                      1                1
   7:                      1                1
   8:                      1                1
   9:                      1                1
 ALL:            310   1   1                1
```

2. In the Voice Paging Timeout field, type **30**.

   This field sets the maximum number of seconds a page can last. In our example, the paging party will be disconnected after 30 seconds.

3. In the Port field for Zone 1, type **01C0501**.

   Use this field to assign a port on an auxiliary trunk circuit pack to this zone.

4. In the Voice Paging — TAC field type **301**.

   Use this field to assign the trunk access code users dial to page this zone. You cannot assign the same trunk access code to more than one zone.

5. In the Voice Paging — COR field type **1**.

   Use this field to assign a class of restriction to this zone. You can assign different classes of restriction to different zones.

6. On the Zone 1 row, type **Reception area** in the Location field.

   Give each zone a descriptive name so you can easily remember the corresponding physical location.

7. Repeat steps 4 through 6 for zones 2 to 5.
8. In the ALL row, type **310** in the Voice Paging — TAC field and **1** in the Voice Paging — COR field.

   By completing this row, you allow users to page all zones at once. You do not have to assign a port to this row.
9. Press ENTER to save your changes.

You can integrate loudspeaker voice paging and call parking. This is called "deluxe paging." You enable deluxe paging by entering y in the Deluxe Paging and Call Park Timeout to Originator field on the Feature-Related System Parameters screen. To allow paged users the full benefit of deluxe paging, you should also enter a code in the Answer Back Access Code field on the Feature Access Code screen if you haven't already: paged users will dial this code + an extension to retrieve calls parked by deluxe paging.

Fixing problems

Problem	Possible causes	Solutions
Users report that they can't page.	The attendant has taken control of the trunk group.	Deactivate attendant control.
Calls to an extension are heard over the loudspeakers.	The extension may have been forwarded to a trunk access code used for paging.	Deactivate call forwarding or change the extension calls are forwarded to.

More information

Users page by dialing the trunk access code assigned to a zone and speaking into their handset. For your users' convenience, you may also want to consider the following options:

- Add the paging trunk access codes to an abbreviated dialing list and allow users to page using the list.
- Assign individual trunk access codes to Autodial buttons.
- Assign individual trunk access codes to Busy buttons. The status lamp tells the user whether or not the trunk is busy.
- For attendants, you can provide one-button paging access by assigning trunk access codes for paging zones to the Direct Trunk Group Select buttons on the attendant console.

With an appropriate class of restriction, remote callers can also make loudspeaker pages.

When deluxe paging is enabled, if a user with an active call dials the trunk access code for a paging zone the active call is automatically parked.

- Users dial the trunk access code + “#” to page and park an active call on their own extensions.
- Users with console permission can park a call on any extension by dialing the trunk access code + the extension.
- Attendants or users with console permissions may park calls to common shared extensions.
- Parked calls can be retrieved from any phone. Paged users simply dial the answer back feature access code + the extension where the call is parked.

Related topics

Refer to ‘‘Paging over speakerphones’’ on page 469 for another way to let users page.

Refer to ‘‘Loudspeaker paging’’ on page 1633 for detailed information on voice paging over loudspeakers.

Setting up chime paging over loudspeakers

Use this procedure to allow users to make chime pages over an external loudspeaker system connected to your switch. Users page by dialing a trunk access code and the extension of the person they want to page. The system plays a unique series of chimes assigned to that extension. This feature is also known as Code Calling Access.

Before you start

Your switch must have one or more auxiliary trunk circuit packs with enough available ports to support the number of paging zones you define. Each paging zone requires 1 port. Refer to the *Avaya MultiVantage™ Hardware Solutions Guide* for information on specific circuit packs.

Instructions

To set up chime paging, you fill out the necessary fields on the Loudspeaker Paging screen and then assign chime codes to individual extensions on the Code Calling IDs screen.

As an example, let's set up chime paging for a clothing store with 3 zones. We'll allow users to page all zones at once, and we'll assign a class of restriction of 1 to all zones.

1. Type **change paging loudspeaker** and press RETURN.

   The Loudspeaker Paging screen appears.

```
                              LOUDSPEAKER PAGING

                                    CDR? y
            Voice Paging Timeout (sec):
           Code Calling Playing Cycles: 2

PAGING PORT ASSIGNMENTS
                 Voice Paging     Code Calling
 Zone   Port       TAC  COR  TN     TAC  COR  TN      Location:
   1:   01A0301              1      80    1   1       Men's Department
   2:   01A0302              1      81    1   1       Women's Department
   3:   01A0303              1      82    1   1       Children's
   4:                        1                1
   5:                        1                1
   6:                        1                1
   7:                        1                1
   8:                        1                1
   9:                        1                1
 ALL:                        1      89    1   1
```

2. In the Code Calling Playing Cycles field, type **2**.

   This field sets the number of times a chime code plays when someone places a page.

3. In the Port field for Zone 1, type **01A0301**.

   Use this field to assign a port on an auxiliary trunk circuit pack to this zone.

4. In the Code Calling — TAC field type **80**.

   Use this field to assign the trunk access code users dial to page this zone. You cannot assign the same trunk access code to more than one zone.

5. In the Code Calling — COR field type **1**.

   Use this field to assign a class of restriction to this zone. You can assign different classes of restriction to different zones.

6. On the Zone 1 row, type **Men's Department** in the Location field.

   Give each zone a descriptive name so you can easily remember the corresponding physical location.

7. Repeat steps 4 through 6 for zones 2 and 3.

8. In the ALL row, type **89** in the Code Calling — TAC field and **1** in the Code Calling — COR field.

   By completing this row, you allow users to page all zones at once. You do not have to assign a port to this row.

9. Press ENTER to save your changes.

To assign chime codes to individual extensions:

1. Type **change paging code-calling-ids** and press RETURN.

   The Code Calling IDs screen appears.

```
                                  CODE CALLING IDs
ID ASSIGNMENTS
         Id   Ext           Id   Ext      Id   Ext      Id   Ext      Id   Ext
         111: 2130_____     141: _______  221: _______  251: _______  331: _______
         112: 2131_____     142: _______  222: _______  252: _______  332: _______
         113: 2149_____     143: _______  223: _______  253: _______  333: _______
         114: 2150_____     144: _______  224: _______  254: _______  334: _______
         115: 2152_____     145: _______  225: _______  255: _______  335: _______
         121: 2153_____     151: _______  231: _______  311: _______  341: _______
         122: 2160_____     152: _______  232: _______  312: _______  342: _______
         123: 2167_____     153: _______  233: _______  313: _______  343: _______
         124: _________     154: _______  234: _______  314: _______  344: _______
         125: _________     155: _______  235: _______  315: _______  345: _______
         131: _________     211: _______  241: _______  321: _______  351: _______
         132: _________     212: _______  242: _______  322: _______  352: _______
         133: _________     213: _______  243: _______  323: _______  353: _______
         134: _________     214: _______  244: _______  324: _______  354: _______
         135: _________     215: _______  245: _______  325: _______  355: _______
```

2. Type the first extension, **2130**, in the Ext field for Id 111.

   Each code Id defines a unique series of chimes.

3. Assign chime codes to the remaining extensions by typing an extension number on the line following each code Id.

   You can assign chime codes to as many as 125 extensions.

4. Press ENTER to save your changes.

Fixing problems

Problem	Possible causes	Solutions
Users report that they can't page.	The attendant has taken control of the trunk group.	Deactivate attendant control.

More information

Users page by dialing the trunk access code assigned to a zone. For your users' convenience, you may also want to consider the following options:

- Add the paging trunk access codes to an abbreviated dialing list and allow users to page using the list.

  NOTE:
  Don't use special characters in abbreviated dialing lists used with chime paging.

- Assign individual trunk access codes to Autodial buttons.
- Assign individual trunk access codes to Busy buttons. The status lamp tells the user whether or not the trunk is busy.
- For attendants, you can provide one-button paging access by assigning trunk access codes for paging zones to the Direct Trunk Group Select buttons on the attendant console.

With an appropriate class of restriction, remote callers can also make loudspeaker pages.

Related Topics

Refer to ''Paging over speakerphones'' below for another way to let users page.

Refer to ''Loudspeaker paging'' on page 1633 for detailed information on chime paging over loudspeakers,

Paging over speakerphones

Use this procedure to allow users to make an announcement over a group of digital speakerphones. By dialing a single extension that identifies a group, users can page over all the speakerphones in that group. Speakerphone paging is one-way communication: group members hear the person placing the page but cannot respond directly.

Before you start

You must have 6400-, 7400-, or 8400-series speakerphones to use speakerphone paging.

Instructions

To set up speakerphone paging, you create a paging group and assign phones to it. In the following example, we'll create paging group 1 and add 4 members.

1. Type **add group-page 1** and press RETURN.

   The Group Paging Using Speakerphone screen appears.

```
                       GROUP PAGING USING SPEAKERPHONE
   Group Number: 1                                    Group Extension: 3210
     Group Name: Sales staff                                       COR: 5
GROUP MEMBER ASSIGNMENTS
    Ext      Name                               Ext       Name
 1: 2009     B. Smith                      17:
 2: 2010     R. Munoz                      18:
 3: 2011     Y. Lu                         19:
 4: 2012     A. Sullivan                   20:
 5:                                        21:
 6:                                        22:
 7:                                        23:
 8:                                        24:
 9:                                        25:
10:                                        26:
11:                                        27:
12:                                        28:
13:                                        29:
14:                                        30:
15:                                        31:
16:                                        32:
```

2. In the Group Extension field, type **3210**.

   This field assigns the extension users dial to page the members of this group.

3. In the Group Name field, type **Sales staff**.

   This name appears on callers' phone displays when they page the group.

4. In the COR field, type **5**.

   Any user who wants to page this group must have permission to call COR 5.

5. In the Ext field in row 1, type **2009**.

6. Enter the remaining extensions that are members of this group.

   The switch fills in the Name fields with the names from the Station screen when you save your changes.

7. Press ENTER to save your changes.

Fixing problems

Problem	Possible causes	Solutions
Users get a busy signal when they try to page.	All phones in the group are busy or off-hook.	Wait a few minutes and try again.
	All phones in the group have Send All Calls or Do Not Disturb activated.	Group members must deactivate these features in order to hear a page.
Some group members report that they don’t hear a page.	Some phones in the group are busy or off-hook.	Wait a few minutes and try again.
	Some phones in the group have Send All Calls or Do Not Disturb activated.	Group members must deactivate these features in order to hear a page.

More information

- You can create up to 32 paging groups on one Avaya MultiVantage system.
- Each group can have up to 32 extensions in it.
- One phone can be a member of several paging groups.

Related topics

Refer to ‘‘Group paging’’ on page 1570 for detailed information on paging over speakerphones.

Paging users who are on active calls

Use this procedure to allow one user to interrupt another user’s call and make a private announcement. This is called whisper paging. The paging user dials a feature access code or presses a feature button, then dials the extension they want to call. All 3 users can hear the tone that signals the page, but only the person on the paged extension can hear the pager’s voice: other parties on the call cannot hear it, and the person making the page cannot hear anyone on the call.

Before you start

Before you administer whisper paging:

- Your switch must have a circuit pack that supports whisper paging. Refer to the *Avaya MultiVantage™ Hardware Solutions Guide* for information on specific models.
- Users must have 6400-, 7400-, 8400-, or 9400-series DCP (digital) phones.

Instructions

You give users the ability to use whisper paging by administering feature buttons or feature access codes.

You can give users feature buttons that make, answer, or block whisper pages. Using the Station screen, you can administer these buttons in any combination as appropriate:

- Whisper Page Activation — allows this user to place a whisper page
- Answerback — allows this user to answer a whisper page

  Pressing the answerback button automatically puts any active call on hold and connects the paged user to the paging user.
- Whisper Page Off— allows this user to block whisper pages

  If possible, assign this function to a button with a lamp so the user can tell when blocking is active. You cannot administer this button to a soft key.

To allow users to make a whisper page by dialing a feature access code, you simply need to enter a code in the Whisper Page Activation Access Code field on the Feature Access Code screen.

Related topics

Refer to ‘‘Using phones as intercoms’’ on page 473 for another way to give individual users quick, two-way communication.

Refer to ‘‘Whisper paging’’ on page 1806 for detailed information on whisper paging.

Using phones as intercoms

Use this procedure to make communications quicker and easier for users who frequently call each other. With the intercom feature, you can allow one user to call another user in a predefined group just by pressing a couple of buttons. You can even administer a button that always calls a predefined extension when pressed.

Instructions

Administering the intercom feature is a 2-step process. First, you create an intercom group and assign extensions to it. Then, to allow group members to make intercom calls to each other, you administer feature buttons on their phones for automatic intercom, dial intercom, or both. This section also provides instructions for allowing one user to pick up another user's intercom calls.

In this example, we'll create intercom group 1 and add extensions 2010 to 2014.

1. Type **add intercom-group 1** and press RETURN.

   The Intercom Group screen appears.

```
                                    INTERCOM GROUP
                                                  Group Number: 1
                                           Length of Dial Code: 1

GROUP MEMBER ASSIGNMENTS
              Ext       DC    Name
           1: 2010      1     B. Smith
           2: 2011      2     L. Yu
           3: 2012      3     R. Munoz
           4: 2013      4     K. Mancetti
           5: 2014      9     N. Mitchell
           6:
           7:
           8:
           9:
          10:
          11:
          12:
          13:
          14:
          15:
          16:
```

2. Type **1** in the Length of Dial Code field.

   Dial codes can be 1 or 2 digits long.

3. On row 1, type **2010** in the Ext field.

4. On row 1, type **1** in the DC field.

   This is the code a user will dial to make an intercom call to extension 2010. The length of this code must exactly match the entry in the Length of Dial Code field.

5. Repeat steps 3 and 4 for the remaining extensions.

   Dial codes don’t have to be in any order. The switch fills in the Name field with the name from the Station screen when you save your changes.

6. Press ENTER to save your changes.

To allow users to make intercom calls, you must administer feature buttons on the phones in the intercom group. You can administer buttons for dial intercom, automatic intercom, or both on multi-appearance phones. You can’t administer either intercom feature on single-line phones, but you can assign single-line phones to intercom groups so those users can receive intercom calls.

As an example, let’s set up automatic intercom between extensions 2010 (dial code = 1) and 2011 (dial code = 2) in intercom group 1.

To set up automatic intercom between extensions 2010 and 2011:

1. Type **change station 2010** and press RETURN.

   The Station screen appears.

```
                                    STATION
 SITE DATA
       Room: _______                                      Headset? n
       Jack: _____                                        Speaker? n
      Cable: _____                                       Mounting: d
      Floor: _______                                  Cord Length: 0
   Building: _______                                    Set Color: ______


ABBREVIATED DIALING
     List1: ________          List2: ________          List3: ________

BUTTON ASSIGNMENTS
  1: call-appr_                        6: __________
  2: call-appr_                        7: __________
  3: call-appr_                        8: __________
  4: auto-icom_  Grp: 1  DC: 2         9: __________
  5: dial-icom_  Grp: 1               10: __________
```

2. Move to the page with the Button Assignments fields.

3. In Button Assignments field 4, type **auto-icom** and press TAB.

   The Grp and DC fields appear.

4. In the Grp field, type **1**.

   This is the number of the intercom group. Since an extension can belong to more than one intercom group, you must assign a group number to intercom buttons.

5. In the DC field, type **2**.

   This is the dial code for extension 2011, the destination extension.

6. Press ENTER to save your changes.

7. Repeat steps 1–6 for extension 2011.

   Assign a dial code of **1** to 2011’s automatic intercom button.

To give a member of a group the ability to make intercom calls to all the other members, administer a Dial Intercom button on the member’s phone. Type the number of the intercom group in the Grp field beside the Dial Intercom button.

You can also give one user instant, one-way access to another. For example, to give user A instant, one-way access to user B, administer an Automatic Intercom button on A’s phone only. You don’t have to administer any intercom button on B’s phone. If B has a Dial Intercom button, he can make an intercom call to A the same way as he would to any other group member.

When users are in the same call pickup group, or if Directed Call Pickup is enabled on your switch, one user can answer an intercom call to another user. To allow users to pick up intercom calls to other users, you must enter y in the Call Pickup on Intercom Calls field on the Feature-Related System Parameters screen.

Related topics

Refer to ‘‘Abbreviated Dialing’’ on page 1333 for information on another way for users to call each other without dialing complete extension numbers.

Refer to ‘‘Intercom’’ on page 1604 for detailed information on intercom functions.

Setting up automatic answer intercom calls

Automatic Answer Intercom Calls (Auto Answer ICOM) allows a user to answer an intercom call within the intercom group without pressing the intercom button. Auto Answer ICOM works with digital, BRI, and hybrid phones with built-in speaker, headphones, or adjunct speakerphone.

SECURITY ALERT:

Press the Do Not Disturb button or the Send All Calls button on your phone when you don't want someone in your intercom group to listen in on a call. Auto Answer ICOM does not work when the Do Not Disturb button or the Send All Calls button is pressed on the phone.

Administration

This section contains an example, with step-by-step instructions, on how to set up Auto Answer ICOM.

In this example, you set up Auto Answer ICOM on station 12345. To do so, complete the following steps:

1. Type **change station 12345**.

   The Station screen for extension 12345 appears.

```
change station 12345                                                 Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? msa-spe            Auto Select Any Idle Appearance? n
          LWC Activation? y                           Coverage Msg Retrieval? y
  LWC Log External Calls? n                                     Auto Answer: icom
             CDR Privacy? n                                 Data Restriction? n
   Redirect Notification? y                      Idle Appearance Preference? n
 Per Button Ring Control? n
   Bridged Call Alerting? n                        Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n           Per Station CPN - Send Calling Number? _
       Service Link Mode: as-needed   Special Character for Restricted Number? n
         Multimedia Mode: basic
    MWI Served User Type: ______                 Display Client Redirection? n
              AUDIX Name:                       Select Last Used Appearance? n
   Messaging Server Name: ______                   Coverage After Forwarding? _
     Recall Rotary Digit? n                          Multimedia Early Answer? n
                                             Direct IP-IP Audio Connections? n
                                                       IP Audio Hairpinning? n
```

2. Move to the Auto Answer field and enter **icom**.
3. Press ENTER to save your changes.

Observing calls

Use this procedure to allow designated users, normally supervisors, to listen to other users' calls. This capability is often used to monitor service quality in call centers and other environments where employees serve customers over the phone. On Avaya MultiVantage, this is called "service observing" and the user observing calls is the "observer."

This section describes service observing in environments without Automatic Call Distribution (ACD) or call vectoring. To use service observing in those environments, refer to *Avaya MultiVantage™ Call Center Software Release 11 Guide to ACD Call Centers*.

Before you start

On the System Parameter Customer-Options screen, verify the:

- Service Observing (Basic) field is y.

If you want to enable remote service observing by allowing remote users to dial a feature access code, verify the:

- Service Observing (Remote/By FAC) field is y

If the appropriate field is not enabled, contact your Avaya representative.

Instructions

SECURITY ALERT:

Listening to someone else's calls may be subject to federal, state, or local laws, rules, or regulations. It may require the consent of one or both of the parties on the call. Familiarize yourself with all applicable laws, rules, and regulations and comply with them when you use this feature.

In this example, we'll set up service observing for a manager. The manager's class of restriction is 5. We'll assign a feature button to the manager's phone and allow her to monitor calls on local extensions that have a class of restriction of 10. Everyone on an observed call will hear a repetitive warning tone.

To set up service observing:

1. Set the observer's class of restriction to permit service observing:
   a. In the Class of Restriction screen for COR 5, type **y** in the Can Be A Service Observer? field.
   b. Move to the page of the Class of Restriction screen that shows service observing permissions.
   c. Type **y** in the field for class of restriction 10.
2. In the Class of Restriction screen for COR 10, type **y** in the Can Be Service Observed? field.

   Anyone with class of restriction 5 now has permission to observe extensions with class of restriction 10. To further restrict who can observe calls or be observed, you may want to create special classes of restriction for both groups and use these classes only for the appropriate extensions.
3. In the Station screen, assign a Service Observing button to the observer's phone.

   A service observing button permits users to switch between listen-only and listen-and-talk modes simply by pressing the button.
4. To activate the warning tone, type **y** in the Service Observing — Warning Tone field on the Feature-Related System Parameters screen.

   A unique 2-second, 440-Hz warning tone plays before an observer connects to the call. While the call is observed, a shorter version of this tone repeats every 12 seconds.

In order for users to activate service observing by feature access codes, use the Feature Access Code screen to administer codes in one or both of the following fields:

- Service Observing Listen Only Access Code
- Service Observing Listen/Talk Access Code

When using feature access codes, observers must choose a mode at the start of the session. They cannot switch to the other mode without ending the session and beginning another.

NOTE:
Feature access codes are required for remote observing.

Related topics

Refer to ''Service observing'' on page 1697 for detailed information on service observing.

Managing data calls

15

Types of data connections

You can use MultiVantageMultiVantage to allow the following types of data elements/devices to communicate to the world:

- Data Terminals
- Personal computers
- Host Computers (for example, CentreVu CMS or Intuity AUDIX)
- Digital Phones (Digital Communications Protocol (DCP) and Integrated Services Digital Network-Basic Rate Interface (ISDN-BRI))
- Audio/Video Equipment
- Printers
- Local Area Networks (LAN)

You enable these connections using a large variety of data communications equipment, such as:

- Modems
- Data Modules
- Asynchronous Data Units (ADUs)
- Modem Pools
- Data/modem pooling circuit packs

Once you have connected these data devices to the switch, you can use networking and routing capabilities to allow them to communicate with other devices over your private network or the public network.

This section describes the system features available to enable data communications.

Data Call Setup

Data Call Setup provides multiple methods to set up a data call:

- Data-terminal (keyboard) dialing
- Telephone dialing
- Hayes AT command dialing
- Administered connections
- Hotline dialing

Administering Data Call Setup

For data-terminal dialing:

1. Choose one of the following data modules and administer all fields:
   - Processor/Trunk Data Module
   - Data Line Data Module
   - 7500 Data Module
2. On the Modem Pool Group screen, administer the Circuit Pack Assignments field.

   Refer to ''Modem Pool Group'' on page 986 for more information.

For telephone dialing:

1. Choose one of the following:
   - On the Feature Access Code screen, administer the Data Origination Access Code field. Refer to ''Feature Access Code'' on page 752 for more information.
   - On the Station screen, assign one button as data-ext (Ext:). Refer to ''Station'' on page 1066 for more information.
2. Choose one of the following data modules and administer all fields:
   - Processor/Trunk Data Module
   - Data Line Data Module

3. On the Modem Pool Group screen, administer the Circuit Pack Assignments field. Refer to ‘‘Modem Pool Group’’ on page 986 for more information.

Depending on the hardware used, assign ports to the following:

- Data modules
- 7400D-series or CALLMASTER digital telephones
- 7500D-series telephones with asynchronous data module (ADM)
- Analog modems (port is assigned using 2500 telephone screen)

Characters used in Data Call Setup

Basic-digit dialing is provided through an ADM or 7500B data module. The user can enter digits from 0 to 9, *, and # from a 7500 or 8500 series telephone keypad or an EIA-terminal interface. In addition, the user can dial the following special characters.

Table 5. Special characters

Character	Use
SPACE, -, (, and)	improves legibility. The switch ignores these characters during dialing.
+ character (wait)	interrupts or suspends dialing until the user receives dial tone
, (pause)	inserts a 1.5-second pause
% (mark)	indicates digits for end-to-end signaling (touch-tone). This is required when the trunk is rotary. It is not required when the trunk is touch-tone.
UNDERLINE or BACKSPACE	corrects previously typed characters on the same line
@	deletes the entire line and starts over with a new DIAL: prompt

Each line of dialing information can contain up to 42 characters (the + and % characters count as two each).

Examples of dialing are:

- DIAL: 3478
- DIAL: 9+(201) 555-1212
- DIAL: 8, 555-2368
- DIAL: 9+555-2368+%9999+123 (remote access)

The following call-progress messages and their meanings are provided for DCP and ISDN-BRI modules.

Table 6. Call-progress messages

Message	Application	Meaning
DIAL:	DCP	Equivalent to dial tone. Enter the desired number or FAC followed by Enter.
CMD	BRI	Equivalent to dial tone. Enter the desired number or FAC followed by Enter.
RINGING	DCP, BRI	Equivalent to ringing tone. Called terminal is ringing.
BUSY	DCP, BRI	Equivalent to busy tone. Called number is busy or out of service.
ANSWERED	DCP, BRI	Call is answered.
ANSWERED - NOT DATA	DCP	Call is answered and a modem answer tone is not detected.
TRY AGAIN	DCP, BRI	Equivalent to reorder tone. System facilities are currently not available.
DENIED	DCP, BRI	Equivalent to intercept tone. Call cannot be placed as dialed.
ABANDONED	DCP, BRI	Calling user has abandoned the call.
NO TONE	DCP, BRI	Tone is not detected.
CHECK OPTIONS	DCP, BRI	Data-module options are incompatible.
XX IN QUEUE	DCP, BRI	Current position in queue.
PROCESSING	DCP, BRI	Out of queue. Facility is available.
TIMEOUT	DCP, BRI	Time is exceeded. Call terminates.

Continued on next page

Table 6. Call-progress messages (Continued)

Message	Application	Meaning
FORWARDED	DCP, BRI	Equivalent to redirection-notification signal. Called terminal activates Call Forwarding and receives a call, and call is forwarded.
INCOMING CALL	DCP, BRI	Equivalent to ringing.
INVALID ADDRESS	DCP	Entered name is not in alphanumeric-dialing table.
WRONG ADDRESS	BRI	Entered name is not in alphanumeric-dialing table.
PLEASE ANS-	DCP, BRI	Originating telephone user transferred call to data module using One-Button Transfer to Data.
TRANSFER	DCP	Data Call Return-to-Voice is occurring.
CONFIRMED	DCP, BRI	Equivalent to confirmation tone. Feature request is accepted, or call has gone to a local coverage point.
OTHER END	DCP, BRI	Endpoint has terminated call.
DISCONNECTED	DCP, BRI	Call is disconnected.
WAIT	DCP, BRI	Normal processing continues.
WAIT, XX IN QUEUE	DCP	Call is in a local hunt-group queue.

DCP data modules

Data-terminal dialing

DCP data-terminal dialing allows a user to set up and disconnect data calls directly from a data terminal as follows.

1. At the DIAL: prompt, the user types the data number. The message RINGING displays.
2. If the call is queued, the message WAIT, XX IN QUEUE displays. The queue position XX updates as the call moves up in queue.

3. To originate and disconnect a call, the user presses BREAK. If the terminal does not generate a 2-second continuous break signal, the user can press originate/disconnect on the data module.
4. The user can enter digits at the DIAL: prompt.

Telephone dialing

DCP telephone dialing allows telephone users to originate and control data calls from a telephone.

Users can set up a call using any unrestricted telephone and then transfer the call to a data endpoint.

The primary way to make data calls is with multiappearance telephone data-extension buttons. Assign any administrable feature button as a data-extension button. The data-extension button provides one-touch access to a data module. The number of assigned data-extension buttons per telephone is not limited.

The following options, either alone or combined, permit flexibility in making data calls from a telephone.

- One-Button Transfer to Data

  A user can transfer a call to the associated data module by pressing the data-extension button after the endpoint answers.

- Return-to-Voice

  A user can change the connection from data to voice. The user presses the data-extension button associated with the busy data module. If the user hangs up, the call disconnects. Return of a data call to the telephone implies that the same data call is continued in the voice mode, or transferred to point.

  The Return-to-Voice feature is denied for analog adjuncts.

- Data Call Preindication

  A user, before dialing a data endpoint, can reserve the associated data module by pressing the data-extension button. This ensures that a conversion resource, if needed, and the data module are reserved for the call. Use of Data Call Preindication before 1-button transfer to data is recommended for data calls that use toll-network facilities. Data Call Preindication is in effect until the associated data-extension button is pressed again for a 1-button transfer; there is no time-out.

ISDN-BRI data modules

Data-terminal dialing

A user can set up and disconnect data calls directly from a data terminal without using a telephone as follows:

1. The user presses ENTER a few times. If the CMD: prompt does not appear, the user presses BREAK A + T at the same time, and then presses ENTER.
2. At the CMD: prompt, the user types and presses ENTER.
3. To disconnect, the user enters +++. At the CMD: prompt, the user types end and presses ENTER.

Telephone dialing

To make a data call, an ISDN-BRI phone user presses the data button on the terminal, enters the number on the dial pad, and then presses the data button again.

The following data functions are not available on ISDN-BRI phones:

- One-Button Transfer to Data
- Return-to-Voice
- Data Call Preindication
- Voice-Call Transfer to Data and Data-Call Transfer to Voice

The system handles all presently defined BRI bearer data-call requests. Some capabilities that are not supported by Avaya terminals are provided by non-Avaya terminals. If the switch does not support a capability, a proper cause value returns to the terminal.

BRI terminals receive a cause or reason code that identifies why a call is being cleared. The BRI data module converts certain cause values to text messages for display.

In a passive-bus multipoint configuration, the system supports two BRI endpoints per port, thus doubling the capacity of the BRI circuit pack. When you change the configuration of a BRI from point-to-point to multipoint, the original endpoint does not need to reinitialize. Only endpoints that support service profile identifier (SPID) initialization can be administered in a multipoint configuration.

Analog modems

When a telephone user places a data call with a modem, the user dials the data-origination access code assigned in the system before dialing the endpoint.

Considerations

- A BRI phone cannot call a data terminal, and a data terminal cannot call a BRI phone.

Interactions

- Abbreviated Dialing

  Only 22 of the 24 (maximum) digits in an abbreviated-dialing number are available for keyboard dialing. The remaining two digits must contain the wait indicator for tone detection.

- Call Coverage

  A hunt group made up of data endpoints cannot be assigned a coverage path.

- Call Detail Recording

  CDR records the use of modem pools on trunk calls.

- Call Forwarding All Calls

  Calls received by a data module can be forwarded. Activate Call Forwarding All Calls with data-terminal (keyboard) dialing. If the forwarded-to endpoint is an analog endpoint and the caller is a digital endpoint, modem pooling is activated automatically.

- Pooled Modems with Hunt Groups

  UCD can provide a group of data modules or analog modems for answering calls to connected facilities (for example, computer ports).

- World-Class Tone Detection

  Multiple-line data-terminal dialing is supported if the administered level of tone detection is precise. You can administer tone-detection options. The message that Data Call Setup sends to users varies according to the option.

  If the option is not set to precise, and a data call is set up over an analog trunk, messages describing the status of the called endpoint (for example, RINGING, BUSY, TRY AGAIN) change according to which tone-detection option is selected.

Default Dialing

Default Dialing provides data-terminal users who dial a specific number the majority of the time a very simple method of dialing that number. Normal data terminal dialing and alphanumeric dialing are unaffected.

Default Dialing enhances data terminal (keyboard) dialing by allowing a data-terminal user to place a data call to a preadministered destination by either pressing a Return at the DIAL: prompt (for data terminals using DCP data modules) or typing **d** and pressing RETURN at the CMD: prompt (for data terminals using ISDN-BRI data modules). The data-terminal user with a DCP data module can place calls to other destinations by entering the complete address after the DIAL: prompt (normal data terminal dialing or alphanumeric dialing). The data-terminal user with an ISDN-BRI data module can place calls to other destinations by typing **d**, a space, the complete address, and press RETURN after the CMD: prompt.

NOTE:
DU-type hunt groups connecting the system to a terminal server on a host computer have hunt-group extensions set to no keyboard dialing.

For the AT command interface supported by the 7400A/7400B/8400B data module, to dial the default destination, enter the ATD command (rather than press return).

Administering Default Dialing

1. You can use an abbreviated dialing list for your default ID. Refer to ‘‘Abbreviated Dialing’’ on page 1333 for more information.
2. On the Data Module screen, administer the following fields:
   - Special Dialing Option as default.
   - Abbreviated Dialing List, enter the list to use.
   - AD Dial Code.

Alphanumeric Dialing

Alphanumeric Dialing enhances data-terminal dialing by allowing users to place data calls by entering an alphanumeric name rather than a long string of numbers.

For example, a user could type 9+1-800-telefon instead of 9+1-800-835-3366 to make a call. Users need to remember only the alpha-name of the far-end terminating point.

Alphanumeric Dialing allows you to change a mapped string (digit-dialing address) without having to inform all users of a changed dial address. Users dial the alpha name.

When a user enters an alphanumeric name, the system converts the name to a sequence of digits according to an alphanumeric-dialing table. If the entered name is not found in the table, the system denies the call attempt and the user receives either an `Invalid Address` message (DCP) or a `Wrong Address` message (ISDN-BRI).

Because data terminals access the switch via DCP or ISDN-BRI data modules, dialing procedures vary:

- For DCP, at the `DIAL`: prompt users type the alphanumeric name and press RETURN.
- For ISDN-BRI, at the `CMD:` prompt users type **d**, a space, and the alphanumeric name, and press RETURN.

More than one alphanumeric name can refer to the same digit string.

Administering Alphanumeric Dialing

1. On the Alphanumeric Dialing Table screen, administer the Alpha-name and Mapped String fields. Refer to ‘‘Alphanumeric Dialing Table’’ on page 585 for more information.

Considerations

NOTE:
Alphanumeric dialing does not apply to endpoints with Hayes modems.

Data Hotline

Data Hotline provides for automatic-nondial placement of a data call preassigned to an endpoint when the originating switch goes off-hook. Use for security purposes.

Administering Data Hotline

1. You can use an abbreviated dialing list for your default ID. Refer to ''Abbreviated Dialing'' on page 1333 for more information.
2. On the Station screen, administer the following fields. Refer to ''Station'' on page 1066 for more information.
   - Abbreviated Dialing List
   - Special Dialing Option
   - Hot Line Destination
3. On the Data Module screen, administer the Abbreviated Dialing List1 field.

The system automatically places Data Hotline calls to preassigned extensions or off-premises numbers. Calling terminals are connected to the system by a data module. Users should store the destination number in the abbreviated dialing list for future reference.

Interactions

- Call Forwarding — All Calls

  A Data Hotline caller cannot activate both Call Forwarding and Data Hotline. Dialing the Call Forwarding feature access code (FAC) causes activation of the Data Hotline instead.

Data Privacy

Data Privacy protects analog data calls from being disturbed by any of the system's overriding or ringing features.

Administering Data Privacy

1. Choose either of the following:
   - On the Feature Access Code screen, administer the Data Privacy Access Code field. Refer to ''Feature Access Code'' on page 752 for more information.
   - On the Class of Service screen, administer the Data Privacy field. Refer to ''Class of Service'' on page 650 for more information.
2. On the Station screen, administer the Class of Service field. Refer to ''Station'' on page 1066 for more information.

To activate this feature, the user dials the activation code at the beginning of the call.

Considerations

- Data Privacy applies to both voice and data calls. You can activate Data Privacy on Remote Access calls, but not on other incoming trunk calls. Data Privacy is canceled if a user transfers a call, is added to a conference call, is bridged onto a call, or disconnects from a call. You can activate Data Privacy on calls originated from attendant consoles.
- For virtual extensions, assign the Data Privacy Class of Service to the mapped-to physical extension.

Interactions

- Attendant Call Waiting and Call Waiting Termination

  If Data Privacy is active, Call Waiting is denied.
- Bridged Call Appearance — Single-Line Telephone

  If you activate Data Privacy or assign Data Restriction to a station involved in a bridged call and the primary terminal or bridging user attempts to bridge onto the call, this action overrides Data Privacy and Data Restriction.
- Busy Verification

  Busy Verification cannot be active when Data Privacy is active.

- Intercom — Automatic and Dial

  An extension with Data Privacy or Data Restriction active cannot originate an intercom call. The user receives an intercept tone.

- Music-on-Hold Access

  If a user places a call with Data Privacy on hold, the user must withhold Music-on-Hold to prevent the transmission of tones that a connected data service might falsely interpret as a data transmission.

- Priority Calls

  If a user activates Data Privacy, Priority Calls are denied on analog telephones. However, Priority Calls appear on the next available line appearance on multiappearance telephones.

Data Restriction

Data Restriction protects analog-data calls from being disturbed by any of the system's overriding or ringing features or system-generated tones.

Data Restriction applies to both voice and data calls.

Once you administer Data Restriction for an analog or multiappearance telephone or trunk group, the feature is active on all calls to or from the terminal or trunk group.

NOTE:
Do not assign Data Restriction to attendant consoles.

Administering Data Restriction

1. On the Station screen, set the Data Restriction field to y. Refer to ''Station'' on page 1066 for more information.
2. Choose one of the following trunk groups and set the Data Restriction field to y. Refer to ''ISDN trunk group'' on page 894 and ''Trunk Group'' on page 1167 for more information.
   - Access
   - Advanced Private-Line Termination (APLT)
   - Circuit Pack (CP)
   - Customer-Premises Equipment (CPE)
   - Direct Inward Dialing (DID)
   - Foreign Exchange (FX)

- Integrated Services Digital Network-Primary Rate Interface (ISDN-PRI)
- Release-Link Trunk (RLT)
- Tandem
- Tie
- Wide Area Telecommunications Service (WATS)

Interactions - Data Restriction

- Attendant Call Waiting and Call Waiting Termination

  If Data Restriction is active, Call Waiting is denied.
- Busy Verification

  Busy Verification cannot be active when Data Restriction is active.
- Intercom — Automatic and Dial

  An extension with Data Privacy or Data Restriction activated cannot originate an intercom call. The user receives an Intercept tone.
- Music-on-Hold Access

  If a user places a call with Data Restriction on hold, The user must withhold Music-on-Hold to prevent the transmission of tones that a connected data service might falsely interpret as a data transmission.
- Priority Calls

  Priority Calls are allowed if the analog station is idle. Call Waiting (including Priority Call Waiting) is denied if the station is busy. However, Priority Calls appear on the next available line appearance on multiappearance telephones.
- Service Observing

  A data-restricted call cannot be service observed.

Data-Only Off-Premises Extensions

Data-Only Off-Premises Extensions allows users to make data calls involving data communications equipment (DCE) or digital terminal equipment (DTE) located remotely from the system site.

A Data-Only Off-Premises Extension uses an on-premises modular trunk data module (MTDM). The system communicates with remote data equipment through the private-line facility linking the on-premises MTDM and the remote data equipment.

Users can place data calls to this type of data endpoint using Telephone Dialing or Data Terminal (Keyboard) Dialing. Since there is no telephone at the remote site, originate data calls from the remote data terminal using Keyboard Dialing only.

Administering Data-Only Off-Premises Extensions

1. On the Processor/Trunk Data Module screen, administer all fields.

   Refer to ‘‘Data modules’’ on page 679 for more information.

Considerations

- The system does not support communications between two TDMs. Modem Pooling is similar to a TDM, it cannot be used on calls to or from a Data-Only Off-Premises Extension.

Interactions

- Telephone Dialing

  An on-premises multiappearance telephone may have a Data Extension button associated with the TDM used for a Data-Only Off-Premises Extension. The telephone user and the remote user share control of the data module. Actions of the user at the telephone may affect the remote user.

  — 1-Button Transfer to Data

  The telephone user can transfer a call to the Data-Only Off-Premises Extension. The Data Extension button lamp on the telephone lights and the Call in Progress lamp on the data module lights during a data call.

  — Data Call Preindication

  The multiappearance telephone user presses the idle associated Data Extension button to reserve a data module. The data module is busy to all other users. When the user reserves a data module, the lamp associated with the Data Extension button winks and lights at any other associated telephones. A remote user receives the BUSY message when attempting to originate a call.

  — Return-to-Voice

  To establish a data call, the telephone user presses the associated busy Data Extension button to transfer the call to the telephone. The data module associated with the Data Extension button is disconnected from the call. The Call in Progress lamp on the data module goes dark.

Data Modules — general

A Data Module is a connection device between a basic-rate interface (BRI) or digital-communications protocol (DCP) interface of the switch and data-terminal equipment (DTE) or data-communications equipment (DCE).

The following types of data modules can be used with the system:

- Announcement data module
- Data line data module
- Processor/trunk data module (P/TDM)
- Netcon data module (DEFINITY Server SI configurations only) Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.
- Processor interface data module (DEFINITY Server SI configurations only). Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.
- System port data module (DEFINITY Server R configurations only)
- X.25 data module (DEFINITY Server R configurations only). Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.
- 7500 data module
- World Class BRI data module
- Ethernet data module. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.
- Point-to-Point Protocol (PPP) data module. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

NOTE:
The 51X series Business Communications Terminals (BCTs) are not administered on the Data Module screen. The 510 BCT (equivalent to a 7405D with a display and built-in DTDM), 515 BCT (equivalent to a 7403D integrated with 7405D display module function, data terminal and built-in DTDM), and the 7505D, 7506D, and 7507D have a DCP interface but have built-in data module functionality. Both are administered through the Station screen.

Detailed description of data modules

TTI allows data modules without hardware translation to merge with an appropriate data module connected to an unadministered port. The unadministered port is given TTI default translation sufficient to allow a terminal connected to the data module (connected to the port) to request a TTI merge with the extension of a data module administered without hardware translation.

NOTE:
TTI is not useful for Announcement and X.25 hardware.

Administration Without Hardware supports PDM, TDM, Data-Line, Announcement, and X.25 data modules.

NOTE:
The 513 BCT has an EIA interface rather than a DCP interface (no built in data module, attachable telephone, or telephone features). The 513 BCT is not administered; only the data module to which the 513 BCT is connected is administered.

7400A/7400B+/8400B+ Data Module

Use the 7400A data module instead of an MTDM when you support combined Modem Pooling. The 7400A data module supports asynchronous operation at speeds up to 19200-bps, and provides a DCP interface to the switch and an EIA 232C interface to the associated modem. The 7400A operates in stand-alone mode as a data module.

7400B+ and 8400B+ data modules support asynchronous-data communications and operate in stand-alone mode for data-only service or in linked mode, which provides simultaneous voice and data service. The 7400B+ and 8400B+ provide voice and data communications to 7400D series phones and 602A1 CALLMASTER phones that have a connection to a data terminal or personal computer. The data modules integrate data and voice into the DCP protocol required to interface with the switch via a port on a digital-line circuit pack. Use the 7400B+ or 8400B+ instead of an MPDM when you need asynchronous operation at speeds up to 19.2-kbps to provide a DCP interface to the switch for data terminals and printers. The 7400B+ and 8400B+ do not support synchronous operation and keyboard dialing. Dialing is provided using the standard Hayes command set.

7400D

This data module supports synchronous operation with AUDIX, CMS, and DCS. It provides synchronous data transmissions at speeds of 19.2-Kbps full duplex.

7400C High Speed Link

The 7400C high-speed link (HSL) is a data-service unit that allows access to DCP data services. It provides synchronous data transmission at speeds of 56- and 64-Kbps and provides a link to high-speed data networks. Used for Group 4 fax applications that include electronic mail and messaging, and electronic storage of printed documents and graphics. Use the 7400C for video teleconferencing and LAN interconnect applications.

7500 Data Modules

The 7500 Data Module connects data-terminal equipment (DTE) or data-communications equipment (DCE) to the ISDN network. The 7500 Data Module supports EIA 232C and V.35 interfaces and RS-366 automatic-calling unit interface (for the EIA 232C interface only).

The 7500 has no voice functions. Configure in the following ways:

- Asynchronous DCE

  300, 1200, 2400, 4800, 9600, 19200-bps
- Synchronous DCE

  1200, 2400, 4800, 9600, 19200, 56000, 64000-bps
- Asynchronous DTE (used for modem pooling)

  up to 19200-bps

The 7500 Data Module is stand-alone or in a multiple-mount housing.

Asynchronous Data Module

NOTE:
The alias station command cannot be used to alias data modules.

Use the Asynchronous Data Module (ADM) with asynchronous DTEs as a data stand for the 7500 and 8500 Series of ISDN-BRI phones, thus providing connection to the ISDN network. The ADM provides integrated voice and data on the same phone and supports data rates of 300, 1200, 2400, 4800, 9600, and 19200-bps. This module also supports the Hayes command set, providing compatibility with PC communications packages.

Related topics

Refer to ‘‘Data modules’’ on page 679 for more information.

Administered Connection

An Administered Connection (AC) is a connection between two access or data endpoints. Avaya MultiVantage automatically establishes and maintains the connection based on the attributes you administer. ACs provides the following capabilities.

- Support of both permanent and scheduled connections
- Auto Restoration (preserving the active session) for connections routed over Software Defined Data Network (SDDN) trunks
- Administrable retry interval (from 1 to 60 minutes) per AC
- Administrable alarm strategy per AC
- Establishment/retry/auto restoration order based on administered priority

Detailed description

Establish an AC between the following:

- Two endpoints on the same switch
- Two endpoints in the same private network, but on different switches
- One endpoint on the controlling switch and another endpoint off the private network

In all configurations, administer the AC on the switch having the originating endpoint. For an AC in a private network, if the two endpoints are on two different switches, normally the connection routes via Automatic Alternate Routing (AAR) through tie trunks (ISDN, DS1, or analog tie trunks) and intermediate switches. If required, route the connection via Automatic Route Selection (ARS) and Generalized Route Selection (GRS) through the public network. The call routes over associated ISDN trunks. When the far-end answers, a connection occurs between the far-end and the near-end extension in the `Originator` field on the AC screen.

Because the system makes an administered connection automatically, you do not use the following:

- Data Call Setup

  Do not assign a default dialing destination to a data module when it is used in an AC.
- Data Hotline

  Do not assign a hotline destination to a data module that is used in an AC.
- Terminal Dialing

  Turn off terminal dialing for data modules involved in an AC. This prevents display of call-processing messages (INCOMING CALL,...) on the terminal.

Access endpoints

Access endpoints are non-signaling trunk ports. They neither generate signaling to the far-end of the trunk nor respond to signaling from the far-end. Designate an access endpoint as the originating endpoint or destination endpoint in an AC.

Typical AC applications

The following are typical AC applications:

- A local data endpoint connection to a local or remote-access endpoint. Examples: an MPDM ACCUNET digital service connecting to SDDN via an ISDN trunk-group DS1 port; an MPDM ACCUNET digital service connecting to an ACCUNET Switched 56 Service via a DS1 port.
- A local-access endpoint connecting to a local or remote-access endpoint. Examples: a DSO cross-connect and a 4-wire leased-line modem to a 4-wire modem connection via an analog tie trunk.
- A local data endpoint connecting to a local or remote data endpoint such as a connection between two 3270 data modules.

NOTE:
The following guidelines do not include AAR and ARS, or GRS administration information for routing AC calls over trunk groups. Refer to the respective feature elsewhere in this book for that information.

Establishing Administered Connections

The originating switch attempts to establish an AC only if one of the following conditions exist:

- AC is active.
- AC is due to be active (either a permanent AC or time-of-day requirements are satisfied if it is a scheduled AC).
- Originating endpoint is in in-service or idle state.

If the originating endpoint is not in service or is idle, no activity takes place for the AC until the endpoint transitions to the desired state. The originating switch uses the destination address to route the call to the desired endpoint. When the switch establishes two or more ACs at the same time, the switch arranges the connections in order of priority.

AC attempts can fail for the following reasons:

- Resources are unavailable to route to the destination.
- A required conversion resource is not available.
- Access is denied by class of restriction (COR), facilities restriction level (FRL), or bearer capability class (BCC). Or, an attempt is made to route voice-band-data over SDDN trunks in the public switch network.
- Destination address is incorrect.
- Destination endpoint is busy.
- Other network or signaling failures occur.

In the event of a failure, an error is entered into the error log, which generates an alarm, if it is warranted by your alarming strategy. You can display AC failures via the status-administered connection command.

As long as an AC is due to be active, the originating switch continues to establish an AC unless the attempt fails because of an administrative error (for example, a wrong number) or service-blocking condition (for example, outgoing calls barred).

- The frequency with which failed attempts are retried is determined by the administered retry interval (1 to 60 minutes) for each AC.
- Retries are made after the retry interval has elapsed regardless of the restorable attribute of the AC.
- ACs are retried in priority order.
- When you change the time of day on the switch, an attempt is made to establish all ACs in the waiting-for-retry state.

Dropping Administered Connections

An AC remains active until one of the following occurs:

- The AC is changed, disabled, or removed.
- The time-of-day requirements of a scheduled AC are no longer satisfied.
- One of the endpoints drops the connection. This could be because of user action (in the case of a data endpoint), maintenance activity resulting from an endpoint failure, busying out of the endpoint, or handshake failure. If the endpoints are incompatible, the connection is successful until handshake failure occurs.

  NOTE:
  An AC between access endpoints remains connected even if the attached access equipment fails to handshake.

- An interruption (for example, facility failure) occurs between the endpoints.

If an AC drops because it was disabled/removed or is no longer due to be active, no action is taken. If an AC drops because of changed AC attributes, an immediate attempt is made to establish the connection with the changed attributes if it is still due to be active. Existing entries in the error/alarm log are resolved if they no longer apply. If handshake failure resulted in the dropping of the connection, in the case of an AC involving at least one data endpoint, no action is taken for that AC until the change administered-connection command is executed.

Administered Connections failure: Auto Restoration and Fast Retry

When an active AC drops prematurely, you must invoke either auto restoration or fast retry to determine whether auto restoration is attempted for an active AC.

If you option AC for auto restoration and the connection was routed over SDDN trunks, auto restoration is attempted. During restoration, connections are maintained between the switch and both endpoints. In addition to allowing the active session to be maintained, AC also provides a high level of security by prohibiting other connections from intervening in active sessions. Auto restoration generally completes before the 60-second endpoint holdover interval. If auto restoration is successful, the call might be maintained (no guarantee). The restoration is transparent to the user with the exception of a temporary disruption of service while restoration is in progress. A successful restoration is reflected by the *restored* state on the status AC screen. Although the restoration was successful, the data session may not have been preserved.

If auto restoration is not active or if the AC is not routed over SDDN trunks, the switch immediately attempts to reestablish the connection (fast retry). The switch also attempts a retry if the originating endpoint initiated the drop. With fast retry, connections are not maintained on both ends. Fast Retry is not attempted for an AC that was last established via fast retry, unless the AC is active for at least two minutes.

If auto restoration or fast retry fails to restore or reestablish the connection, the call drops and the AC goes into retry mode. Retry attempts continue, at the administered retry interval, as long as the AC is due to be active.

Administering Administered Connections

1. Choose one of the following data modules and administer all fields:
   - Data Line Data Module (use with Data Line circuit pack)
   - Processor/Trunk Data Module (use with one of the following:)
     — MPDMs, 700D, 7400B, 7400D, or 8400B
     MTDMs, 700B, 700C, 700E, or 7400A
   - Processor Interface Data Module (refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information)
   - X.25 Data Module (refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information)
   - 7500 Data Module (use with ISDN Line 12-BRI-S-NT or ISDN Line 12-BRI-U-NT circuit pack)
   - World Class Core BRI Data Module (use with wcbri)
2. On the DS1 Circuit Pack screen, administer all fields. Refer to ''DS1 Circuit Pack'' on page 728 for more information. (Use with switch node carriers.)
3. On the Access Endpoint screen, administer all fields. Refer to ''Access Endpoint'' on page 575 for more information.
4. Choose one of the following trunk groups and administer all fields. Refer to ''ISDN trunk group'' on page 894 and ''Trunk Group'' on page 1167 for more information.
   - ISDN-BRI
   - ISDN-PRI
   - Tie
5. On the Class of Restriction screen, administer all fields. Refer to ''Class of Restriction'' on page 636 for more information.
6. On the Class of Service screen, administer all fields. Refer to ''Class of Service'' on page 650 for more information.
7. On the Dial Plan Record screen, administer the Local Node Number field with a number from 1-63 that matches the DCS switch node number and the CDR node number. Refer to ''DCS to QSIG TSC Gateway screen'' on page 724 for more information.
8. On the Administered Connection screen, administer all fields. Refer to ''Administered Connection'' on page 579 for more information.

9. On the Station screen, assign one button as ac-alarm. Refer to ''Station'' on page 1066 for more information.
10. On the Attendant Console screen, assign one button as ac-alarm. Refer to ''Attendant Console'' on page 595 for more information.

Interactions

- Abbreviated Dialing

  Use Abbreviated Dialing entries in the `Destination` field. Entries must comply with restrictions.

- Busy Verification of Stations and Trunks

  This feature does not apply to access endpoints because they are used only for data.

- Call Detail Recording

  For an AC that uses a trunk when CDR is active, the origination extension is the originator of the call.

- Class of Restriction

  Reserve a COR for AC endpoints and SDDN trunks. This restricts endpoints that are not involved in AC from connecting to SDDN trunks or endpoints involved in AC.

- Class of Service/Call Forwarding

  Assign to an AC endpoint a COS that blocks Call Forwarding activation at the endpoint.

- Digital Multiplexed Interface (DMI)

  Use DMI endpoints as the destination in an AC. DMI endpoints do not have associated extensions, so do not use them as the originator in an AC.

- Facility Test Calls

  The feature does not apply to access endpoints because an access endpoint acts as an endpoint rather than as a trunk.

- Modem Pooling

  If you require a modem in an AC, one is inserted automatically. If no modem is available, the connection is dropped.

- Non-Facility Associated Signaling (NFAS) and D-Channel Backup

  Auto restoration for an AC that is initially routed over an NFAS facility may fail if the only backup route is over the facility on which the backup D-channel is administered. The backup D-channel may not come into service in time to handle the restoration attempt.

- Set Time Command

  When you change the system time via the set time command, all scheduled ACs are examined. If the time change causes an active AC to be outside its scheduled period, the AC is dropped. If the time change causes an inactive AC to be within its scheduled period, the switch attempts to establish the AC.

  If any AC (scheduled or continuous) is in retry mode and the system time changes, the switch attempts to establish the AC.

- System Measurements

  Access endpoints are not measured. All other trunks in an AC are measured as usual.

Modem Pooling

Modem Pooling allows switched connections between digital-data endpoints (data modules) and analog-data endpoints via pods of acoustic-coupled modems. The analog-data endpoint is either a trunk or a line circuit.

Data transmission between a digital data endpoint and an analog endpoint requires conversion via a modem, because the DCP format used by the data module is not compatible with the modulated signals of an analog modem. A modem translates DCP format into modulated signals and vice versa.

Modem Pooling feature provides pools of integrated-conversion modems and combined-conversion modems.

Integrated-conversion modem pools have functionality integrated on the Pooled Modem circuit pack, providing two modems. Each one emulates a TDM cabled to a 212 modem. Integrated are modem pools not available in countries that use A-law companding.

Combined-conversion modem pools are TDMs cabled to any TDM-compatible modem. Combined-conversion modem pools can be used with all systems.

The system can detect the needs for a modem. Data calls from an analog-data endpoint require that the user indicate the need for a modem, because the system considers such calls to be voice calls. Users indicate this need by dialing the data-origination access code field on the Feature Access Code screen before dialing the digital-data endpoint.

The system provides a Hold Time parameter to specify the maximum time any modem can be held but not used (while a data call is in queue).

Administering Modem Pooling

For Integrated modem poolings:

1. On the Modem Pool Group screen, administer all fields. Refer to ''Modem Pool Group'' on page 986 for more information.
2. On the Feature Access Code screen, administer the Data Origination Access Code field. Refer to ''Feature Access Code'' on page 752 for more information.
3. On the Data Module screen, administer all fields. Refer to ''Data modules'' on page 679 for more information.

For Combined modem poolings:

1. On the Modem Pool Group screen, administer all fields. Refer to ''Modem Pool Group'' on page 986 for more information.
2. On the Feature Access Code screen, administer the Data Origination Access Code field. Refer to ''Feature Access Code'' on page 752 for more information.

Considerations

- On data calls between a data module and an analog-data endpoint, Return-to-Voice releases the modem and returns it to the pool. The telephone user connects to the analog-data endpoint.
- For traffic purposes, the system accumulates data on modem-pooling calls separate from voice calls. Measurements on the pools also accumulate.
- Modem Pooling is not restricted. Queuing for modems is not provided, although calls queued on a hunt group retain reserved modems.
- Avoid mixing modems from different vendors within a combined pool because such modems may differ in transmission characteristics.
- Each data call that uses Modem Pooling uses four time slots (not just two). As a result, heavy usage of Modem Pooling could affect TDM bus-blocking characteristics.
- Tandem switches do not insert a pooled modem. The originating and terminating switches insert a pooled modem.

Interactions

- Call Detail Recording

  Data call CDR records the use of modem pools on trunk calls.

- Data Call Setup

  Data calls to or from a TDM cannot use Modem Pooling.

- Data Privacy and Data Restriction

  The insertion of a modem pool does not turn off Data Privacy or Data Restriction.

- Data-Only Off-Premises Extensions

  Calls to or from a Data-Only Off-Premises Extension cannot use Modem Pooling, when this type of digital-data endpoint uses a TDM.

- DMI Trunks

  If you place a data call from a local analog-data endpoint to a DMI trunk, you must dial the data-origination access code to obtain a modem. Data calls on DMI trunks to local analog-data endpoints automatically obtain modems.

- DS1 Tie Trunk Service

  Connect modems used for Modem Pooling to AVD DS1 tie trunks via Data Terminal Dialing or by dialing the feature-access code for data origination.

PC Interface

The personal computer (PC) Interface consists of the PC/PBX platforms and PC/ISDN Platform product family. These products are used with Avaya MultiVantage to provide users of IBM-compatible PCs fully-integrated voice and data workstation capabilities.

Two groups of different configurations are available for PC Interface: group 1 uses Digital Communications Protocol (DCP) and group 2 uses the ISDN-BRI (Basic Rate Interface) protocol.

The group 1 configurations consist of DCP configurations that use a DCP expansion card) in the PC to link to the switch. Group 1 (shown in Figure 12 on page 506) uses the following connections:

- The PC Interface card plugs into an expansion slot on the PC. The card has 2 standard 8-pin modular jacks (line and phone).

- The digital phone plugs into the phone jack on the PC Interface card.

- The line jack on the card provides a digital port connection to Avaya DEFINITY Servers.
- The distance between the PC Interface card and the PBX should be no more than 1524m for 24-gauge wire or 1219m for 26-gauge wire.

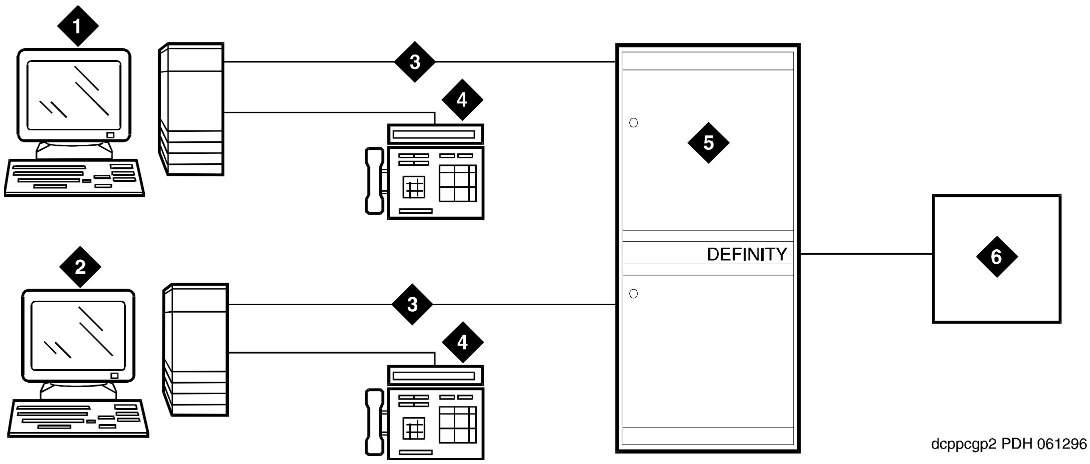


Figure Notes

1. IBM-compatible PC with DCP Interface card
2. IBM-compatible PC with DCP Interface card
3. DCP
4. DCP telephone
5. Avaya (Digital Line, Digital Line (16-DCP-2-Wire), or Digital Line (24-DCP-2-wire) circuit pack)
6. Host

Figure 12. DCP PC interface configuration (Group 1)

The group 2 configurations link to the switch using a PC/ISDN Interface card installed in the PC. This group can include a stand-alone PC terminal, or up to 4 telephones, handsets, or headsets. Group 2 (shown in Figure 13 on page 507) uses PC/ISDN Interface cards (up to four cards) which plug into expansion slots on the PC. These cards each provide 2 standard 8-pin modular-jack connections for both line connections (to the switch) and phone connections. A standard 4-pin modular jack is also available for use with a handset or headset.

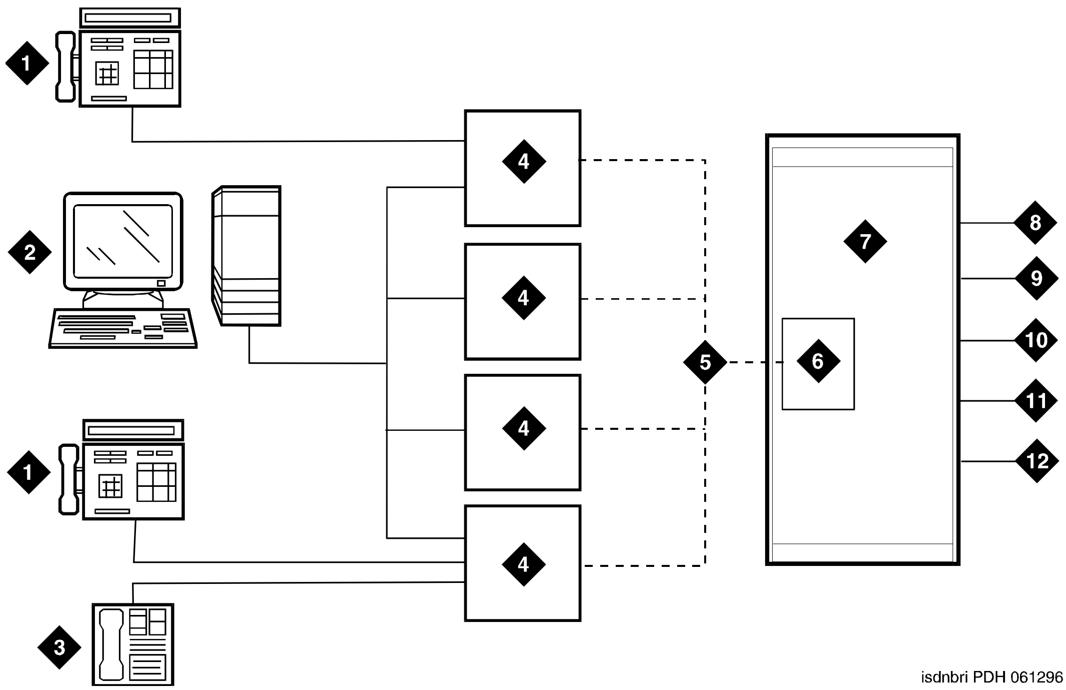


Figure Notes

1. ISDN telephone
2. PC with application
3. Handset or Headset
4. BRI Interface card
5. 2B + D
6. ISDN Line (12-BRI-S-NT) circuit pack)
7. Avaya Media Server
8. PRI trunks
9. BRI stations
10. Interworking
11. DMI
12. Switch features

Figure 13. ISDN—BRI PC interface configuration (Group 2)

PC Interface users have multiple appearances (depending on the software application used) for their assigned extension. Designate one or more of these appearances for use with data calls. With the ISDN-BRI version, you can use up to 4 separate PC/ISDN Interface cards on the same PC. Assign each card a separate extension, and assign each extension one or more appearances. The availability of specific features depends on the COS of the extension and the COS for the switch. Modem Pooling is provided to ensure general availability of off-net data-calling services.

Security

There are two areas where unauthorized use may occur with this feature: unauthorized local use and remote access.

SECURITY ALERT:

Unauthorized local use involves unauthorized users who attempt to make calls from a PC. The PC software has a security setting so users can place the PC in Security Mode when it is unattended. You also can assign Automatic Security so that the administration program on the PC is always active and runs in Security Mode. This mode is password-protected.

SECURITY ALERT:

Remote access involves remote access to the PC over a data extension. Remote users can delete or copy PC files with this feature. You can password-protect this feature. Refer to the Avaya Products Security Handbook for additional steps to secure your system and to find out about obtaining information regularly about security developments.

Administering a PC interface

1. On the Station screen, set the Type field to **pc**.

Considerations

- Use the Function Key Module of the 7405D with PC Interface.
- BRI terminals normally are initializing terminals and require you to assign an SPID. The PC/ISDN Platform (Group 2), in a stand-alone configuration, is a non-initializing BRI terminal and does not require you to assign a SPID.
  — Set a locally-defined terminal type with General Terminal Administration
  — Define the terminal type as a non-initializing terminal that does not support Management Information Messages (MIM).
  — Assign the PC/ISDN Platform with an associated (initializing) ISDN-BRI telephone (such as an ISDN 7505) using a SPID.
  — Assign the station (using a locally-defined terminal type) to take full advantage of the capabilities of the PC Interface. This terminal type is also non-initializing with no support of MIMs.

- Do not use telephones with data modules with the PC Interface. (You can still use 3270 Data Modules if you also use 3270 emulation). If you attach a DCP data module or ISDN data module to a telephone that is connected to a PC Interface card, the data module is bypassed (not used). All the interface functions are performed by the interface card even if a data module is present.
- The 7404D telephone with messaging cartridge cannot be used with PC Interface. However, the 7404D with PC cartridge can be used, but only with Group 1 configurations.

Wideband Switching

Wideband Switching provides the ability to dedicate 2 or more ISDN-PRI B-channels or DS0 endpoints for applications that require large bandwidth. It provides high-speed end-to-end communication between endpoints where dedicated facilities are not economic or appropriate. ISDN-BRI trunks do not support wideband switching.

Wideband Switching supports:

- High-speed video conferencing
- WAN disaster recovery
- Scheduled batch processing (for example, nightly file transfers)
- LAN interconnections and imaging
- Other applications involving high-speed data transmission, video transmission, or high bandwidth

Detailed description

ISDN-PRI divides a T1 or E1 trunk into 24 (32 for E1) channels, where one channel is used for signaling, and all others for standard narrowband communication. Certain applications, like video conferencing, require greater bandwidth. You can combine several narrowband channels into one wideband channel to accommodate the extra bandwidth requirement. Avaya MultiVantage serves as a gateway to many types of high-bandwidth traffic. In addition, DS1 Converter circuit packs are used for wideband switching at DS1 remote EPN locations. They are compatible with both a 24-channel T1 and 32-channel E1 facility (transmission equipment). They support circuit-switched wideband connections (NxDS0) and a 192 Kbps packet channel.

The following table provides information on Wideband Switching channel types.

Channel Type	Number of Channels (DSOs)	Data Rate
H0 (T1 or E1)	6 (grouped 4 (T1) or 5 (E1) quadrants of 6 B-channels each)	384 Kbps
H11 (T1 or E1)	24 (on T1 - all 24 B-channels, with the D-channel not used; on E1 - B-channels 1 to 15, and 17 to 25, and B-channels 26 to 31 unused)	1536 Kbps
H12 (E1 only)	30 (B-channels 1 to 15 and 17 to 31)	1920 Kbps
NxDS0 (T1)	2-24	128–1536 Kbps
NxDS0 (E1)	2-31	128–1984 Kbps

Channel allocation

For standard narrowband communication, ISDN-PRI divides a T1 or E1 trunk as follows:

- T1 trunks are divided into 23 information channels are 1 signaling channel
- E1 trunks are divided into 30 information channels, 1 signaling channel, and 1 framing channel

Certain applications, like video conferencing, require greater bandwidth. You can combine several narrowband channels into one wideband channel to accommodate the extra bandwidth requirement. Avaya MultiVantage serves as a gateway to many types of high-bandwidth traffic. In addition, DS1 converters are used for wideband switching at remote locations.

Performed using one of the three allocation algorithms: fixed, flexible, or floating.

- Fixed allocation — Provides contiguous-channel aggregation. The starting channel is constrained to a predetermined starting point. (Used only for H0, H11, and H12 calls.)
- Flexible allocation — Allows a wideband call to occupy non-contiguous positions within a single T1 or E1 facility (NxDS0).
- Floating allocation — Enforces contiguous-channel aggregation. The starting channel is not constrained to a predetermined starting point (NxDS0).

Typical uses

A typical video application uses an ISDN-PRI interface to DS0 1 through 6 of the line-side facility. Refer to Figure 14.

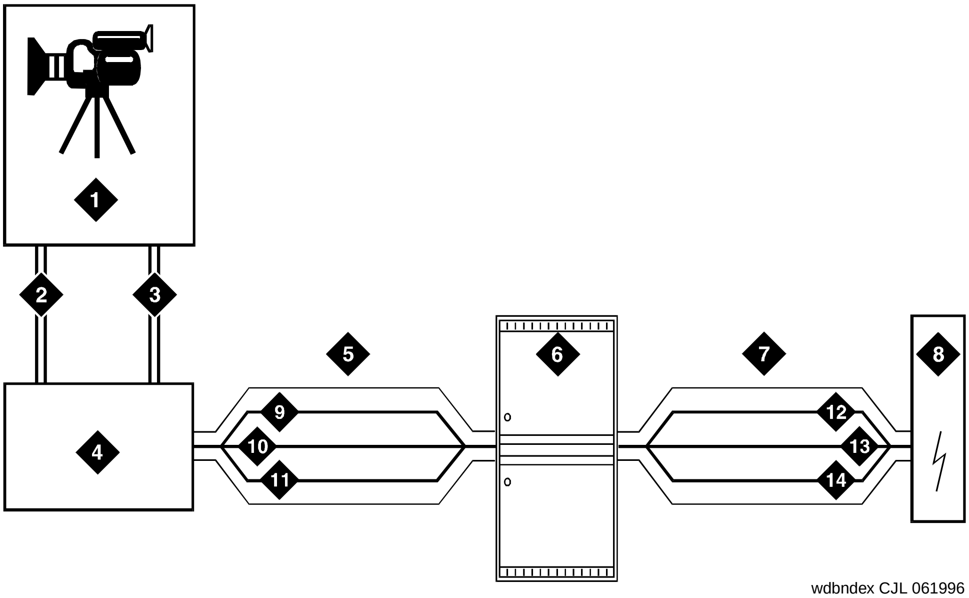


Figure Notes

1. Video application
2. Port 1
3. Port 2
4. ISDN terminal adaptor
5. Line-side ISDN-PRI
6. Avaya Media Server
7. ISDN trunk
8. Network
9. DS0 24 D-channel
10. DS0 23 unused
11. DS0 1–6 wideband
12. DS0 24 D-channel
13. DS0 7–23 narrow bands
14. DS0 1–6 wideband

Figure 14. Wideband Switching Video Application

Endpoint applications

An endpoint application is the origination or destination of a wideband call. Endpoint application can be any number of data applications based on the customer's particular needs.

ISDN-PRI terminal adapters

For wideband switching with non-ISDN-PRI equipment, you can use an ISDN-PRI terminal adapter. ISDN-PRI terminal adapters translate standard ISDN signaling into a form that can be used by the endpoint application and vice versa. The terminal adapter also must adhere to the PRI-endpoint boundaries as administered on the Avaya MultiVantage switch when handling both incoming (to the endpoint) applications and outgoing calls.

Line-side (T1 or E1) ISDN-PRI facility

A line-side ISDN-PRI (T1 or E1) facility is comprised of a group of DS0s (24 for a T1 facility and 32 for an E1 facility). In this context, these DS0s are also called channels. T1 facilities have 23 B-channels and a single D-channel. E1 facilities have 30 B-channels, 1 D-channel, and a framing channel. Data flows bi-directionally across the facility between the switch and the ISDN-PRI terminal adapter.

PRI-endpoints

A PRI-endpoint (PE) is a combination of DS0 B-channels on a line-side ISDN-PRI facility that has been assigned an extension.

A PRI-endpoint can support calls of lower bandwidth. In other words, a PE having a width 6 (six DS0s) can handle a call of one channel (64 Kbps) up to an including 6 channels. For example, an endpoint application connected to a PE defined as using B-channels 1 through 6 of an ISDN-PRI facility could originate a call using B-channels 1, 3, and 5 successfully. If the PE has been administered to use flexible channel allocation, the algorithm for offering a call to the PE starts from the first DS0 administered to the PE. Since only one active call is permitted on a PE, contiguous B-channels always are selected unless one or more B-channels are not in service.

One facility can support multiple separate and distinct PRI-endpoints (several extensions) within a single facility. Non-overlapping contiguous sets of DS0s (B-channels) are associated with each PE.

Universal digital signaling level 1 circuit pack

The UDS1 circuit pack is the interface for line-side and network facilities carrying wideband calls.

Non-signaling configuration

Wideband also can support configurations using non-signaling (non-ISDN-PRI) line-side T1 or E1 connections. The endpoints are the same as those defined for configurations with signaling.

Data service unit/channel service unit

This unit simply passes the call to the endpoint application. Unlike terminal adapters, the DSU/CSU does not have signaling capability.

NOTE:
No DSU/CSU is needed if the endpoint application has a fractional T1 interface.

Line-side (T1 or E1) facility

This facility, like the ISDN-PRI facility, is composed of a group of DS0s (24 for a T1 facility and 32 for an E1 facility; both T1 and E1 use 2 channels for signaling purposes). Line-side facilities are controlled solely from the switch. Through the access-endpoint command, a specific DS0 or group of DS0s is assigned an extension. This individual DS0 or group, along with the extension, is known as a Wideband Access Endpoint (WAE).

Wideband access endpoint

WAEs have no signaling interface to the switch. These endpoints simply transmit and receive wideband data when the connection is active.

NOTE:
The switch can determine if the connection is active, but this does not necessarily mean that data is actually coming across the connection.

A WAE is treated as a single endpoint and can support only one call. If all DS0s comprising a wideband access endpoint are in service, then the wideband access endpoint is considered in service. Otherwise, the wideband access endpoint is considered out of service. If an in-service wideband access endpoint has no active calls on its DS0s, it is considered idle. Otherwise, the wideband access endpoint is considered busy.

Multiple WAEs are separate and distinct within the facility and endpoint applications must be administered to send and receive the correct data rate over the correct DS0s. An incoming call at the incorrect data rate is blocked.

Guidelines and examples

This section examines wideband and its components in relation to the following specific customer usage scenarios:

- Data backup connection
- Scheduled batch processing
- Primary data connectivity
- Networking

Data backup connection

Using wideband for data transmission backup provides customers with alternate transmission paths for critical data in the event of primary transmission path failure.

Scheduled batch processing

Scheduled batch processing applications are used for periodic database updates (for example, retail inventory) or distributions (for example, airline fare schedules). These updates are primarily done after business hours and are often referred to as nightly file transfers. Wideband meets the high bandwidth requirements at low cost for scheduled batch processing. In addition, wideband allows the dedicated-access bandwidth for busy-hour switch traffic to be used for these applications after business hours; no additional bandwidth costs are incurred.

The non-ISDN backup data connection is also appropriate for scheduled batch processing applications. Administered Connections are used to schedule daily or weekly sessions originating from this application.

Primary data connectivity

Permanent data connections (those always active during business hours), such as interconnections between LANs, are well suited for Avaya MultiVantage when ISDN-PRI endpoints are used. The ISDN end-to-end monitoring and the endpoint’s ability to react to failures provide for critical data availability needs. With ISDN, endpoints can detect network failures and initiate backup connections through the switch; ISDN endpoints can also establish additional calls when extra bandwidth is needed.

Any failures not automatically restored by Avaya MultiVantage are signaled to the endpoint application, which can initiate backup data connections over the same PRI endpoint. Avaya MultiVantage routes the backup data connections over alternate facilities if necessary.

Networking

All of the wideband networking is over ISDN-PRI facilities (and the emulation of them by ATM-CES) but may connect to a variety of networks, other domestic interexchange carriers' services, private line, RBOC services, and services in other countries.

ISDN-PRI trunk groups and channel allocation

Only ISDN-PRI trunks (and the emulation of them by ATM-CES) support wideband calls to the network. Wideband's bandwidth requirements have necessitated modification of the algorithms by which trunks look for idle channels. The following section describes the search methods and their relationship to the available wideband data services.

Facility lists

The system always sends a wideband call over a single trunk group and a single DS1 facility (or other ISDN-PRI-capable facility). Since a trunk group may contain channels (trunk members) from several different DS1 facilities, the system maintains a facility list for each trunk group.

A facility list orders the trunk members based on signaling group. If the system is using non-facility associated signaling groups with multiple DS1 facilities, the system sorts trunk members in that signaling group according to the interface identifier assigned to the corresponding DS1 facility.

When searching for available channels for a wideband call placed over a given trunk group, the system starts with the channels in the lowest-numbered signaling group with the lowest interface identifier. If the system cannot find enough channels in a given signaling group with that interface identifier, it checks the next higher interface identifier. If no more interface identifiers are available in the current signaling group, the system moves its search to the channels in the next higher signaling group.

For example, if three facilities having signaling group/interface identifier combinations of 1/1, 1/2, and 2/1 were associated with a trunk group, then a call offered to that trunk group would search those facilities in the order as they were just listed. Also note that since trunks within a given facility can span several trunk groups, a single facility can be associated with several different trunk groups.

Given this facility list concept, the algorithms have the ability to search for trunks, by facility, in an attempt to satisfy the bandwidth requirements of a given wideband call. If one facility does not have enough available bandwidth to support a given call, or it is not used for a given call due to the constraints presented in the following section, then the algorithm searches the next facility in the trunk group for the required bandwidth (if there is more than one facility in the trunk group).

In addition to searching for channels based on facilities and required bandwidth, Port Network (PN) preferential trunk routing is also employed. This PN routing applies within each algorithm at a higher priority than the constraints put on the algorithm by the parameters listed later in this section. In short, all facilities that reside on the same PN as the originating endpoint are searched in an attempt to satisfy the bandwidth of a given call, prior to searching any facilities on another PN.

Direction of trunk/hunting within facilities

You can tell the system to search for available channels in either ascending or descending order. These options help you reduce glare on the channels because the system can search for channels in the opposite direction to that used by the network. If an ISDN trunk group is not optioned for wideband, then a cyclical trunk hunt based on the administration of trunks within the trunk group is still available.

H11

When a trunk group is administered to support H11, the algorithm to satisfy a call requiring 1,536 Kbps of bandwidth uses a fixed allocation scheme. That is, the algorithm searches for an available facility using the following facility-specific channel definitions:

- T1: H11 can only be carried on a facility without a D-channel being signaled in an NFAS arrangement (B-channels 1-24 are used).
- E1: Although the 1,536 Kbps bandwidth could be satisfied using a number of fixed starting points (for example, 1, 2, 3, and so forth), the only fixed starting point being supported is 1. Hence, B-channels 1-15 and 177-25 always are used to carry an H11 call on an E1 facility.

If the algorithm cannot find an available facility within the trunk that meets these constraints, then the call is blocked from using this trunk group. In this case, the call may be routed to a different trunk group preference via Generalized Route Selection (GRS), at which time, based on the wideband options administered on that trunk group, the call would be subject to another hunt algorithm (that is, either the same H11 algorithm or perhaps an N x DS0 algorithm described in a later paragraph).

Note that on a T1 facility, a D-channel is not considered a busy trunk and results in a facility with a D-channel always being partially contaminated. On an E1 facility, however, a D-channel is not considered a busy trunk because H11 and H12 calls may still be placed on that facility; an E1 facility with a D-channel and idle B-channels is considered an idle facility.

H12

Since H12 is 1,920 Kbps which is comprised of 30 B-channels, a 1,920-kbps call can only be carried on an E1 facility. As with H11, the hunt algorithm uses a fixed allocation scheme with channel 1 being the fixed starting point. Hence, an H12 call always is carried on B-channels 1 to 15 and 17 to 31 on an E1 facility (as shown in the following table). When offered any other call (other than a 1,536-kbps call), the algorithm behaves as it does when H11 is optioned.

		DS0s Comprising Each Channel	
Facility	**ISDN Interface**	**H11**	**H12**
T1	23B + D	-	-
T1	24B (NFAS)	1-24	-
E1	30B + D	1-15, 17-25	1-15, 17-31
E1	31B (NFAS)	1-15, 17-25	1-15, 17-31

H0

When a trunk group is administered to support H0, the algorithm to satisfy a call requiring 384 Kbps of bandwidth also uses a fixed allocation scheme. Unlike the H11 fixed scheme which only supports a single fixed starting point, the H0 fixed scheme supports 4 (T1) or 5 (E1) starting points. The H0 algorithm searches for an available quadrant within a facility based on the direction of trunk or hunt administered. If the algorithm cannot find an available quadrant within any facility allocated to this trunk group, then the call is blocked from using this trunk group. Again, based on GRS administration, the call may route to a different trunk group preference and be subject to another algorithm based on the wideband options administered.

Note that a D-channel is considered a busy trunk and results in the top most quadrant of a T1, B-channels 19 to 24, always being partially contaminated. This is *not true* for NFAS.

If this H0 optioned trunk group is also administered to support H11, H12, or N x DS0, then the system also attempts to preserve idle facilities. In other words, when offered a narrowband, H0, or N x DS0 call, the system searches partially-contaminated facilities before it searches to idle facilities.

N x DS0

For the N x DS0 multi-rate service, a trunk group parameter determines whether a floating or a flexible trunk allocation scheme is to be used. The algorithm to satisfy an N x DS0 call is either floating or flexible.

- Floating (Contiguous) — In the floating scheme, an N x DS0 call is placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without any constraint being put on the starting channel (that is, no fixed starting point trunk).
- Flexible — In the flexible scheme, an N x DS0 call is placed on any set of B-channels as long as the requested bandwidth is satisfied. There is absolutely no constraint such as contiguity of B-channels or fixed starting points. Of course, as with all wideband calls, all the B-channels comprising the wideband call must reside on the same ISDN facility.

  Regardless of the allocation scheme employed, the N x DS0 algorithm, like the H11 and H12 algorithms, attempts to preserve idle facilities when offered B, H0, and N x DS0 calls. This is important so that N x DS0 calls, for large values of N, have a better chance of being satisfied by a given trunk group. However, if one of these calls cannot be satisfied by a partially-contaminated facility and an idle facility exists, a trunk on that idle facility is selected, thus contaminating that facility.

  There are additional factors to note regarding specific values of N and the N x DS0 service:

  — N = 1 — this is considered a narrowband call and is treated as any other voice or narrowband-data (B-channel) call.

  — N = 6 — if a trunk group is optioned for both H0 and N x DS0 service, a 384-kbps call offered to that trunk group is treated as an H0 call and the H0 constraints apply. If the H0 constraints cannot be met, then the call is blocked.

  — N = 24 — if a trunk group is optioned for both H11 and N x DS0 service, a 1,536-kbps call offered to that trunk group is treated as an H11 call and the H11 trunk allocation constraints apply.

  — N = 30 — if a trunk group is optioned for both H12 and N x DS0 service, a 1,920-kbps call offered to that trunk group is treated as an H12 call and the H12 trunk allocation constraints apply.

Glare and blocking

Glare prevention

Glare occurs when both sides of an ISDN interface select the same B-channel for call initiation. For example, a user side of an interface selects the B-channel for an outgoing call and, before the switch receives and processes the SETUP message, the switch selects the same B-channel for call origination. Since any single wideband call uses more channels, the chances of glare are greater. With proper and careful administration, glare conditions can be reduced.

To reduce glare probability, the network needs to be administered so both sides of the interface select channels from opposite ends of facilities. This is called linear hunting, ascending or descending. For example, on a 23B+D trunk group, the user side could be administered to select B-channels starting at channel 23 while the network side would be administered to start selecting at channel 1. Using the same example, if channel 22 is active but channel 23 is idle, the user side should select channel 23 for re-use.

Blocking prevention

Blocking occurs when insufficient B-channels required to make a call are available. Narrowband calls require only one channel so blocking is less likely than with wideband calls which require multiple B-channels. Blocking also occurs for wideband calls when bandwidth is not available in the appropriate format (that is, fixed, floating, or flexible).

To reduce blocking, the switch selects trunks for both wideband and narrowband calls to maximize availability of idle fixed channels for H0, H11, and H12 calls and idle floating channels for N x DS0 calls that require a contiguous bandwidth. The strategy for preserving idle channels to minimize blocking depends on the channel type. The chances for blocking are reduced if you use a flexible algorithm, assuming it is supported on the other end.

Channel Type	Blocking Minimization Strategy
H0	Preserve idle quadrants
H11	Preserve idle facilities
H12	Preserve idle facilities
Flexible NxDS0	Preserve idle facilities
Floating NxDS0	Preserve idle facilities as first priority

Administering Wideband Switching

Before you start

You need a DS1 Converter circuit pack. Refer to the Avaya MultiVantage *System Description* for more information on the circuit pack.

Instructions

To administer wideband switching:

1. On the Access Endpoint screen, administer all fields.

   Refer to ''Access Endpoint'' on page 575 for more information.
2. On the PRI Endpoint screen, administer all fields.

   Refer to ''PRI Endpoint'' on page 1024 for more information.
3. On the ISDN Trunk Group screen, administer all fields.

   Refer to ''ISDN trunk group'' on page 894 for more information.
4. On the Route Pattern screen, administer all fields.

   Refer to ''Route Pattern'' on page 1039 for more information.

   NOTE:
   The following is optional.
5. On the Fiber Link Administration, administer all fields.

   Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

Considerations

- For wideband switching with non-ISDN-PRI equipment, you can use an ISDN-PRI terminal adapter.

Interactions

- Administered Connections

  Provides call initiation for Wideband Access Endpoints (WAEs). All Administered Connections that originate from WAEs use the entire bandwidth administered for WAE. The destination of an Administered Connection can be a PRI endpoint.

- Automatic Circuit Assurance

  Treats wideband calls as logical single-trunk calls so that a single ACA-referral call is made if an ACA-referral call is required. The call is referred to the lowest B-channel associated with the wideband call.

- Call Coverage

  A wideband endpoint extension cannot be administered as a coverage point in a call-coverage path.

- Call Detail Recording

  When CDR is active for the trunk group, all wideband calls generate CDR records. The feature flag indicates a data call and CDR records contain bandwidth and Bearer Capability Class (BCC).

- Call Forwarding

  You must block Call Forwarding through Class of Service.

- Call Management System and Basic Call Management System

  Wideband calls can be carried over trunks that are measured by CMS and BCMS. Wideband endpoints are not measured by CMS and BCMS.

- Call Vectoring

  PRI endpoints can use a vector-directory number (VDN) when dialing. For example, PRI endpoint 1001 dials VDN 500. VDN 500 points to Vector 1. Vector 1 can point to other PRI endpoints such as route-to 1002, or route-to 1003, or busy.

  Call Vectoring is used by certain applications. When an incoming wideband call hunts for an available wideband endpoint, the call can route to a VDN, that sends the call to the first available PRI endpoint.

- Class of Restriction

  COR identifies caller and called-party privileges for PRI endpoints. Administer the COR so that account codes are not required. Forced entry of account codes is turned off for wideband endpoints.

- Facility Busy Indication

  You can administer a busy-indicator button for a wideband-endpoint extension, but the button does not accurately track endpoint status.

- Facility Test Calls

  You can use Facility Test Calls to perform loop-back testing of the wideband call facility.

- Generalized Route Selection

  GRS supports wideband BCC to identify wideband calls. GRS searches a route pattern for a preference that has wideband BCC. Route preferences that support wideband BCC also can support other BCCs to allow different call types to share the same trunk group.

- CO Trunk (TTC - Japan) Circuit Pack

  This circuit pack cannot perform wideband switching. No member of the circuit pack should be a member of a wideband group.

CallVisor Adjunct-Switch Application Interface

CallVisor Adjunct-Switch Applications Interface (ASAI) links Avaya MultiVantage and adjunct applications. The interface allows adjunct applications to access switch features and supply routing information to the switch. CallVisor ASAI improves Automatic Call Distribution (ACD) agents' call handling efficiency by allowing an adjunct to monitor, initiate, control, and terminate calls on the switch. The CallVisor ASAI interface may be used for Inbound Call Management (ICM), Outbound Call Management (OCM), and office automation/messaging applications.

CallVisor ASAI is supported by two transport types. These are:

1. Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) transport (CallVisor ASAI-BRI)
2. LAN Gateway Transmission Control Protocol/Internet Protocol transport (Avaya LAN Gateway).

CallVisor ASAI messages and procedures are based on the ITU-T Q.932 international standard for supplementary services. The Q.932 Facility Information Element (FIE) carries the CallVisor ASAI requests and responses across the interface. An application program can access CallVisor ASAI services by supporting the ASAI protocol or by using a third-party vendor application programming interface (API).

For a simple ASAI configuration example, refer to Figure 15.

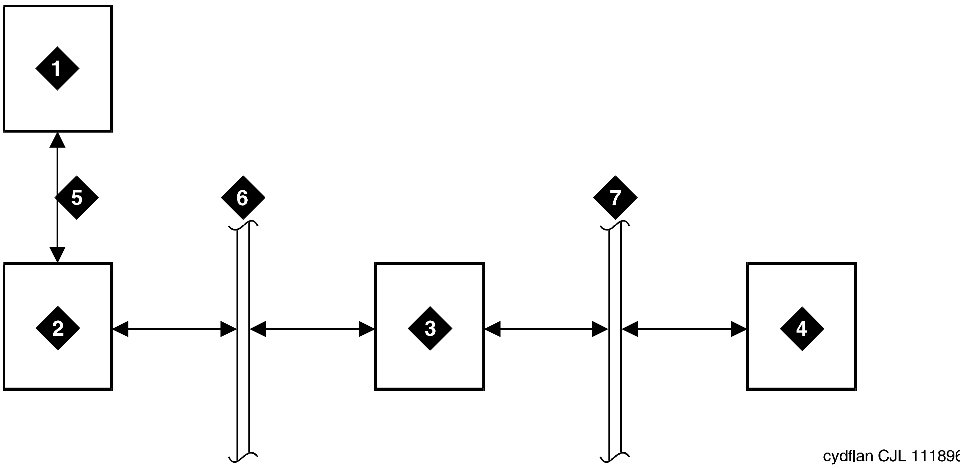


Figure Notes

1. ASAI adjunct
2. ISDN Line circuit pack
3. Packet Controller circuit pack
4. Switch processing element (SPE)
5. ISDN-BRI
6. Packet bus
7. Memory bus

Figure 15. ASAI Switch Interface Link — BRI Transport

ASAI Capabilities

For information concerning the types of associations over which various event reports can be sent, refer to *Avaya MultiVantage™ CallVisor ASAI Technical Reference*.

Considerations

- If your system has an expansion cabinet (with or without duplication), ASAI resources should reside on the system's Processor Cabinet.

Interactions

Refer to *Avaya MultiVantage™ CallVisor ASAI Technical Reference*.

Setting up CallVisor ASAI

CallVisor Adjunct-Switch Applications Interface (ASAI) can be used in the telemarketing and help-desk environments. It is used to allow adjunct applications to monitor and control resources in Avaya MultiVantage.

Before you start

- On the System-Parameters Customer-Options screen, verify the:
  - ASAI Interface field is **y**. If not, contact your Avaya representative.
  - ASAI Proprietary Adjunct Links field is **y** if the adjunct is running the CentreVu Computer Telephony.

Instructions

To set up CallVisor ASAI:

1. Type **add station *nnnn*** and press RETURN, where *nnnn* is the extension you want to assign to the ASAI adjunct.

   The Station screen appears.

```
                                                              Page 1 of X
                                          STATION

Extension: 1014                               Lock Messages? n       BCC: 0
     Type: asai                               Security Code: ______   TN: 1
     Port:                                  Coverage Path 1: ___     COR: 1
     Name: __________________________       Coverage Path 2: ___     COS: 1
                                            Hunt-to-Station: ____

STATION OPTIONS
             Data Module? n                 Personalized Ringing Pattern: 1
          Display Module? n                             Message Lamp Ext: 1014


                                                    MM Complex Data Ext: ________
```

2. In the Type field, type
   - **asai** if this adjunct platform is other than CentreVu Computer Telephony, for example, IBM CallPath.
   - **adjlk** (Computer Telephony adjunct link) if this is for the CentreVu Computer Telephony using the Telephony Services Application Programming Interface (TSAPI).
3. In the Port field, use the port address assigned to the LAN Gateway Interface circuit pack.
4. Press ENTER to save your changes.

Collecting billing information

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Collecting information about calls

Call Detail Recording (CDR) collects detailed information about all incoming and outgoing calls on specified trunk groups. If you use intraswitch CDR, you can also collect information about calls between designated extensions on the switch. Avaya MultiVantage sends this information to a printer or to some other CDR output device that collects call records and that may also provide reports.

You can have a call accounting system directly connected to your switch. If you are recording call details from several switches, the switch may send the records to a collection device for storage. A system called a poller may then take these records and send them to the call accounting system. The call accounting system sorts them, and produces reports that you can use to compute call costs, allocate charges, analyze calling patterns, detect unauthorized calls, and keep track of unnecessary calls.

Before you start

The call accounting system that you use may be sold by Avaya, or it may come from a different vendor. You need to know how your call accounting system is set up, what type of call accounting system or call detail recording unit you are using, and how it is connected to the switch. You also need to know the record format that your call accounting system requires.

Instructions

In this example, we are going to establish call detail recording for all calls that come in on trunk group 1 (our CO trunk). We are going to set up CDR so that any call that is handled by an attendant produces a separate record for the attendant part of the call.

1. Type **change trunk-group 1** and press RETURN.

   The Trunk Group screen appears.

```
                                TRUNK GROUP

Group Number: 1                    Group Type: co          CDR Reports: y
  Group Name: Outside calls               COR: 85      TN: 1__      TAC: 105
   Direction: two-way        Outgoing Display? n
 Dial Access? n                Busy Threshold: 99     Night Service: 1234
Queue Length: 0             Country: 1          Incoming Destination: attd
   Comm Type: voice                  Auth Code? n  Digit Absorption List: _
    Prefix-1? y                    Trunk Flash? n        Toll Restricted? y

TRUNK PARAMETERS
            Trunk Type: ground start
    Outgoing Dial Type: tone                                Cut-Through? n
     Trunk Termination: rc                    Disconnect Timing(msec): 500_

            Auto Guard? n    Call Still Held? n     Sig Bit Inversion: none

                                   Trunk Gain: high

Disconnect Supervision - In? y  Out? n
Answer Supervision Timeout: 10             Receive Answer Supervision? n
```

2. In the CDR Reports field, type **y**.

   This tells the switch to create call records for calls made over this trunk group.

3. Press ENTER to save your changes.

4. Type **change system-parameters cdr** and press RETURN.

   The CDR System Parameters screen appears.

```
                              CDR SYSTEM PARAMETERS
                                                                 Page 1 of X
Node Number (Local PBX ID): 1       CDR Date Format: month/day
        Primary Output Format: ___________       Primary Output Ext: ________
      Secondary Output Format: ___________     Secondary Output Ext: ________
             Use ISDN Layouts? _                EIA Device Bit Rate: ____
         Use Enhanced Formats? _
    Modify Circuit ID Display? _                 Remove # from Called Number? _
                    Record Outgoing Calls Only? _           Intra-switch CDR? _
    Suppress CDR for Ineffective Call Attempts? _     Outg Trk Call Splitting? _
        Disconnect Information in Place of FRL? _       Outg Attd Call Record? _
                                                       Interworking Feat-flag? _
   Force Entry of Acct Code for Calls Marked on Toll Analysis Screen? _
                                      Calls to Hunt Group - Record: __________
  Record Called Vector Directory Number Instead of Group or Member? _
           Record Called Agent Login ID Instead of Group or Member? _
       Inc Trk Call Splitting? _                  Inc Attd Call Record? _
    Record Non-Call-Assoc TSC? _          Call Record Handling Option: _________
        Record Call-Assoc TSC? _   Digits to Record for Outgoing Calls: _________
     Privacy - Digits to Hide: _               CDR Account Code Length: __
```

5. In the CDR Format field, type **month/day**.

   This determines how the date will appear on the header record.

6. In the Primary Output Format field, type **Unformatted**.

   This is the record format that our call accounting system requires. Check with your call accounting vendor to determine the correct record format for your system.

7. In the Primary Output Ext. field, type **2055**.

   This is the extension of the data module that we use to connect to our call accounting system.

8. In the Record Outgoing Calls Only field, type **n**.

   This tells the switch to create records for both incoming and outgoing calls over all trunk groups that use CDR.

9. In the Outg Trk Call Splitting and Inc Trk Call Splitting fields, type **y**.

   This tells the system to create a separate record for any portion of an incoming or outgoing call that is transferred or conferenced.

10. In the Outg Att Call Record and Inc Att Call Record fields, type **y**.

    This tells the system to create a separate record for the attendant portion of any incoming or outgoing call.

More information

You can also administer the switch to produce separate records for calls that are conferenced or transferred. This is called Call Splitting. There are many other variations that you can administer for CDR, and these are described in the screens section of this book.

Related topics

For additional information on CDR, refer to ''Call Detail Recording'' on page 1437.

For more information about how to connect a CDR output device, refer to the *Installation for Adjuncts and Peripherals for Avaya MultiVantage™ Solutions*.

The *Call Detail Acquisition and Processing Reference Manual* also contains CDR information, but has not been updated in several years and is no longer entirely accurate.

Recording calls between users on the same switch

Call detail recording generally records only those calls either originating or terminating outside the switch. There may be times when you need to record calls between local users. Intra-switch CDR lets you track calls made to and from local extensions.

Instructions

In this example, we administer the switch to record all calls to and from extensions 5100, 5101, and 5102.

1. Type **change system-parameters cdr** and press RETURN.

   The CDR System Parameters screen appears.

2. In the Intraswitch CDR field, type **y**.
3. Press ENTER to save your changes.
4. Type **change intra-switch-cdr** and press RETURN.

The Intra-Switch CDR screen appears.

```
                                                                  Page  1  of  1
                                   INTRA-SWITCH CDR
Assigned Members: 2 of 100 administered
 1: 72447 __ 19:________ 37:________ 55:________ 73:________ 91:_______
 2: 72448    20:________ 38:________ 56:________ 74:________ 92:_______
 3: ________ 21:________ 39:________ 57:________ 75:________ 93:_______
 4: ________ 22:________ 40:________ 58:________ 76:________ 94:_______
 5: ________ 23:________ 41:________ 59:________ 77:________ 95:_______
 6: ________ 24:________ 42:________ 60:________ 78:________ 96:_______
 7: ________ 25:________ 43:________ 61:________ 79:________ 97:_______
 8: ________ 26:________ 44:________ 62:________ 80:________ 98:_______
 9:_________ 27:________ 45:________ 63:________ 81:________ 99:_______
10:_________ 28:________ 46:________ 64:________ 82:_______ 100._______
11:_________ 29:________ 47:________ 65:________ 83:_______ 101:_______
12:_________ 30:________ 48:________ 66:________ 84:_______ 102:_______
13:_________ 31:________ 49:________ 67:________ 85:_______ 103:_______
14:_________ 32:________ 50:________ 68:________ 86:_______ 104:_______
15:_________ 33:________ 51:________ 69:________ 87:_______ 105:_______
16:_________ 34:________ 52:________ 70:________ 88:_______ 106:_______
17:_________ 35:________ 53:________ 71:________ 89:_______ 107:_______
18:_________ 36:________ 54:________ 72:________ 90:_______ 108:_______
```

5. In the first three available slots, type **5100**, **5101**, and **5102**.
6. Press ENTER to save your changes.
7. The switch will now produce call records for all calls to and from these extensions, including those that originated on the local switch.

Related topics

See ‘‘Intraswitch CDR’’ on page 1443 for more detailed information.

Tracking calls by account code

You can have your users to enter account codes before they make calls. By doing this, you can have a record of how much time was spent on the phone doing business with or for a particular client.

Instructions

In this example, we are going to set up the system to allow the user at extension 5004 to enter a 5-digit account code before making a call.

1. Type **change system-parameters cdr** and press RETURN.

   The CDR System Parameters screen appears.

```
                        CDR SYSTEM PARAMETERS
                                                              Page 1 of X
   Node Number (Local PBX ID): 1                    CDR Date Format: _________
        Primary Output Format: ___________        Primary Output Ext: ________
Secondary Output Format: ___________      Secondary Output Ext: ________
             Use ISDN Layouts? _                 EIA Device Bit Rate: ____
         Use Enhanced Formats? _
    Modify Circuit ID Display? _                Remove # from Called Number? _
                    Record Outgoing Calls Only? _           Intra-switch CDR? _
    Suppress CDR for Ineffective Call Attempts? _     Outg Trk Call Splitting? _
        Disconnect Information in Place of FRL? _       Outg Attd Call Record? _
                                                     Interworking Feat-flag? _
   Force Entry of Acct Code for Calls Marked on Toll Analysis Screen? _
                                   Calls to Hunt Group - Record: __________
  Record Called Vector Directory Number Instead of Group or Member? _
           Record Called Agent Login ID Instead of Group or Member? _
       Inc Trk Call Splitting? _                Inc Attd Call Record? _
    Record Non-Call-Assoc TSC? _          Call Record Handling Option: _________
        Record Call-Assoc TSC? _    Digits to Record for Outgoing Calls: _________
     Privacy - Digits to Hide: _                CDR Account Code Length: __
```

2. In the CDR Account Code Length field, type **5.**
3. Press ENTER to save your changes.
4. Assign an account button on the Station screen for extension 5004. See ''Adding feature buttons'' on page 127 for more information.
5. Provide your users with a list of account codes to use.
6. You can also assign a Feature Access Code and give this to your users.

Forcing users to enter account codes

Forced Entry of Account Codes is another form of account code dialing. You can use it to allow certain types of calls only with an account code, to track fax machine usage by account, or just to make sure that you get account information on all relevant calls.

Before you start

Before you can administer Forced Entry of Account Codes, it must be enabled on the System Parameters Customer-Options screens. If it is not, please contact your Avaya representative.

Instructions

In this example, we administer the system to force users in our North American office to enter an account code before making international calls.

1. Type **change system-parameters cdr** and press RETURN.

   The CDR System Parameters screen appears.

2. In the Force Entry of Acct Code for Calls Marked on Toll Analysis Form field, type **y**.
3. In the CDR Account Code Length field, type **5.**
4. Press ENTER to save your changes.
5. Type **change toll 0** and press RETURN.
6. The Toll Analysis screen appears.

```
change toll analysis                                                Page 1 of 1
                                TOLL ANALYSIS                     Percent Full:_
                               Location:
                         Total         Toll  CDR  <--Unrestricted Call List-->
    Dialed String       Min  Max  RCL  List  FEAC  1  2  3  4  5  6  7  8  9 10
0_____________________  1__  1__   _     x     _   _  _  _  _  _  _  _  _  _  _
01____________________  9__  17_   _     x     x   _  _  _  _  _  _  _  _  _  _
011___________________  10_  18_   _     x     x   _  _  _  _  _  _  _  _  _  _
______________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
______________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
______________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
______________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
______________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
______________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
______________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
______________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
______________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
```

7. In the first available Dialed String field, type **011**.

   This is the international access code for this office.

8. In the Total Min and Max columns, type 10 and 18, respectively.

   This is the minimum and maximum number of digits the system will analyze when determining how to handle the call.

9. In the Toll List and CDR FEAC columns, type x.
10. Press ENTER to save your changes.

More information

You can also establish a class of restriction with Forced Entry of Account Codes set to **y**, and assign this COR to trunks or other facilities that you want to restrict. With this method, all users with this COR must enter account codes before making any outgoing trunk calls. See ''Class of Restriction'' on page 636 for more information.

Receiving call-charge information

Avaya MultiVantage provides two ways to receive information from the public network about the cost of calls. Note that this service is not offered by the public network in some countries, including the US.

- Advice of Charge (for ISDN trunks) collects charge information from the public network for each outgoing call. Charge advice is a number representing the cost of a call; it may be recorded as either a charging or currency unit.
- Periodic Pulse Metering (for non-ISDN trunks) Periodic Pulse Metering (PPM) accumulates pulses transmitted from the public network at periodic intervals during an outgoing trunk call. At the end of the call, the number of pulses collected is the basis for determining charges.

Before you start

You need to request either AOC or PPM service from your network provider. In some areas, your choice may be limited. Your Avaya representative can help you determine the type of service you need.

NOTE:
This service is not offered by the public network in some countries, including the US.

Collecting call charge information over ISDN

In this example, we administer the system to provide Advice of Charge over an existing ISDN trunk group, at the end of a call. This information will appear on CDR reports.

1. Type **change trunk-group 2**.

   The ISDN trunk group screen appears with existing administration for this ISDN trunk group.

```
                                                                    Page 1 of x
                                 TRUNK GROUP

Group Number: 2                     Group Type: isdn          CDR Reports: y
  Group Name: OUTSIDE_CALL_______________ COR: 1_       TN: 1__       TAC: ____
   Direction: two-way_        Outgoing Display? n
 Dial Access? n               Busy Threshold: 99_       Night Service: ________
Queue Length: 0__
Service Type: public-ntwrk            Auth Code? n           TestCall ITC: rest
                         Far End Test Line No: _______________
TestCall BCC: 4
TRUNK PARAMETERS
         Codeset to Send Display: 6      Codeset to Send TCM,Lookahead: 6
        Max Message Size to Send: 260    Charge Advice: none_____________
  Supplementary Service Protocol: a      Digit Handling(in/out):enbloc/enbloc

            Trunk Hunt: cyclical             QSIG Value-Added Avaya? n
     Connected to Toll? n       STT Loss: normal       DTT to DCO Loss: normal
Calling Number - Delete: ___  Insert: _______________ Numbering Format: _______
              Bit Rate: 1200_       Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0__
```

2. In the CDR Reports field, type **y.**

   This ensures that the AOC information appears on the CDR report.

3. Verify that Service Type is **public-ntwrk**.
4. In the Supplementary Service Protocol field, type **a**.
5. The Charge Advice field, type **end-on-request**.

   This ensures that the switch will place one request for charge information. This reduces the amount of information passed to the switch and consumes less processor time than other options.

6. Press ENTER to save your changes.

Receiving call-charge information over non-ISDN trunks

In this example, we will administer an existing DIOD trunk to receive PPM from the public network.

1. Type **change trunk-group 3**.

   The Trunk Group screen appears with existing administration for this trunk group.

```
                                                                Page 1 of x
                                  TRUNK GROUP

 Group Number: 3                   Group Type: diod          CDR Reports: y
   Group Name: DIOD_PPM___________________ COR: 1_      TN: 1__      TAC: 112_
    Direction: two-way_       Outgoing Display? n
  Dial Access? n              Busy Threshold: 99_       Night Service: ________
 Queue Length: 0__
    Comm Type: voice                    Auth Code? n   Digit Absorption List:
     Prefix-1? y                      Trunk Flash? n         Toll Restricted? y
TRUNK PARAMETERS
             Trunk Type: auto/immed
    Outgoing Dial Type: tone                           Incoming Dial Type: tone
     Trunk Termination: rc
       Digit Treatment:                                           Digits:
       Expected Digits:                               Sig Bit Inversion: none

                                  Trunk Gain: high       Drop Treatment: silence
  Disconnect Supervision - In? y  Out? n
  Answer Supervision Timeout: 0__          Recieve Answer Supervision: y
```

2. In the CDR Reports field, type **y.**

   This ensures that the PPM information appears on the CDR report.

3. In the Direction field, type **two-way**.
4. In the PPM field, type **y**.
5. In the Frequency field, type **50/12**.

   This is the signal frequency (in kHz). The frequency you will use depends on what the circuit pack you use is able to accept. See ‘‘Trunk Group’’ on page 1167 for more information.

6. In the Administrable Timers section, set the Outgoing Glare Guard timer to **5** seconds.
7. Press ENTER to save your changes.
8. You also need to ensure that the values of the Digital Metering Pulse Minimum, Maximum and Value on the DS1 Circuit Pack screen are appropriate to the values offered by your service provider. See ‘‘DS1 Circuit Pack’’ on page 728 for more information.

Related topics

See "Call Charge Information" on page 1411 for more information about AOC and PPM.

Viewing call charge information

Avaya MultiVantage provides two ways for you to view call-charge information: on a telephone display or as part of the Call Detail Recording (CDR) report. From a display, users can see the cost of an outgoing call, both while the call is in progress and at the end of the call.

Instructions

In this example, we administer extension 5040 to be able to view the charge of a call in progress. The charges will appear in currency units (in this case, Lira) on the user's telephone display.

1. Type **change trunk-group 2** and press RETURN.

   The Trunk Group screen appears.

```
add trunk-group next                                              Page 2 of x
TRUNK FEATURES
         ACA Assignment? _      Measured: ____
                                                  Maintenance Tests? _
                        Data Restriction? _

  Abandoned Call Search? _
  Suppress # Outpulsing? _

      Charge Conversion: _____
          Decimal Point: ______
        Currency Symbol: ___
            Charge Type: _______   Receive Analog Incoming Call ID: ________
                                        Per Call CPN Blocking Code: ___
                                      Per Call CPN Unblocking Code: ___
                                                     MF Tariff Free? _
           Outgoing ANI: _______________
```

2. In the Charge Conversion field, type **200**.

   This indicates that one charge unit sent from the service provider is equal to 200 units, in this case, Lira.

3. In the Decimal Point field, type **none**.
4. In the Charge Type field, type **Lira**.
5. Press ENTER to save your changes.

6. Type **change system-parameters features** and press RETURN.

   The Feature-Related System Parameters screen appears.

7. In the Charge Display Update Frequency (seconds) field, type **30**.

   Frequent display updates may have considerable performance impact.

8. Press ENTER to save your changes.

9. Now assign extension 5040 a disp-chrg button to give this user the ability to control the charge display. See ‘‘Adding feature buttons’’ on page 127 for more information.

More information

If you want end users to control when they view this information, you can assign a display button that they can press to see the current call charges. If you want call charges to display automatically whenever a user places an outgoing call, you can set Automatic Charge Display to **y** on the user’s Class of Restriction (COR) screen.

Administering the S8300 Media Server and G700 Media Gateway

17

This chapter describes how to administer an Avaya G700 Media Gateway and the Avaya S8300 Media Server. It is targeted for system administrators after the product is installed and tested. In a converged network where voice and data are both sent over a corporate local area network (LAN), this configuration can provide primary or standby telephony and communications-processing capabilities.

Users should have broad data networking experience with Avaya, Cisco, Nortel, 3Com, or equivalent products to best understand this product. An in-depth knowledge of Avaya MultiVantage call-processing software and Avaya Cajun systems is also recommended.

Overview

To set up and maintain your S8300 Media Server with a G700 Media Gateway you need to administer:

- the G700 Media Gateway and its internal processors, typically using the P330 Device Manager web-based tool or a command-line interface (CLI)
- the S8300 Media Server using the S8300 Media Server Web Interface
- call-processing features using Avaya MultiVantage software

For details administration access methods, tools, and alternatives, see *Welcome to the Avaya S8300 Media Server and Avaya G700 Media Gateway*. Documents mentioned in this chapter are available on the Avaya website: `http://www.avaya.com/support`

Media Gateway administration

The hardware elements of the G700 Media Gateway are summarized in this section. For details on any of these hardware components, or for different configuration setups, see the *Avaya MultiVantage Solutions Hardware Guide*.

Media Gateway physical design

The G700 Media Gateway is a19-inch, rack-mount design similar to other Avaya Cajun hardware. The media gateway can be a member of a Cajun stack of Layer 2 or 3 devices, reside in a stack of other media gateways, or operate as a standalone unit. Figure 16 shows the front of the media gateway.

Figure 16. The G700 Media Gateway (front view)

Key components of the G700 Media Gateway include:

1. LED Status Panel (top left) for the media gateway
2. S8300 Media Server in left center slot (V1), see ''S8300 Media Server faceplate'' below for details
3. Cajun Expansion slot (bottom left) for adding modules such as a 16-port Ethernet expansion module, fiber interfaces, ATM access or WAN access
4. Dual 10/100Base-T Ethernet Ports for connection to the corporate LAN (bottom center of chassis)
5. Up to three additional media modules in slots V2 to V4 on the right side of the chassis (slots are numbered from top to bottom)
6. 8-Pin RS-232 Serial Port (Console) for direct-connect Cajun stack processor administration (typically done by Services personnel)

S8300 Media Server faceplate

Figure 17 shows the faceplate of the S8300 Media server.

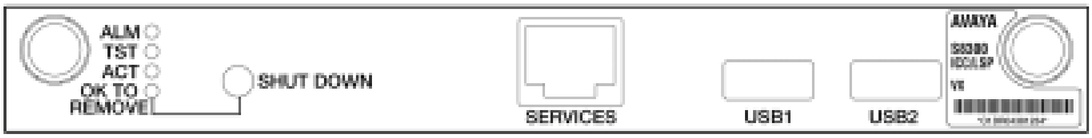

Figure 17. S8300 Media Server faceplate

The faceplate of the S8300 Media Server has the following components:

- The LED array on the left indicates when the S8300 is active and when it is safe to power it down.
- A Shut Down button can be used to shut down or restart the server (the S8300 can also be shut down and restarted using software commands).
- A Services Ethernet interface provides a direct connection to a laptop computer connected with a crossover cable. Normally only Services technicians use this interface; most ongoing S8300 administration is done using the Ethernet connection to the corporate LAN on the media gateway.
- One of the two USB connections may be used to attach an external modem, primarily used to report alarms to a Services support agency.
- The label on the right provides identification information for the S8300 Media Server.

Media module description

The four slots on the G700 Media Gateway are populated with media modules according to the needs at your site. Up to four slots may be filled as follows:

- The S8300 Media Server is a special type of media module that hosts the call-processing (telephony) software. If present, the S8300 is installed in slot one (V1); only one can be installed per media gateway. The S8300 can act as a primary call controller for the system, or one or more S8300s can be configured to provide standby service for a primary call controller if needed. See ‘‘Local Survivable Processor configuration’’ for details.

- The other media module slots are filled as needed to meet the call-processing needs at your site. Media modules can be mixed and matched. Table 7 lists the administrative name and port capacities for the media modules that may be installed in a G700 Media Gateway.

Table 7. Media Module names and port capacities

Media module type	Administration name	Number of ports
Analog Line/Trunk	analog	8
DCP Telephone	dcp	8
T1/E1	ds1	24/32 (T1=24 in robbed-bit signalling mode and E1=32 for 1 control, 1 signal and 30 voice)
VoIP	voip	N/A
S8300 Media Server	icc	N/A

Local Survivable Processor configuration

An S8300 Media Server can be configured either as the primary call-processing controller, or as a Local Survivable Processor (LSP). An LSP can take over call processing if the primary call-processing system (such as another S8300 Media Server or an S8700 Media Server) is unavailable for any reason (such as a reboot or server problem). The primary and LSP media servers can both be S8300 Media Servers; they are set up to operate as a primary or standby LSP server during the configuration process using the S8300 Media Server Web Interface.

If the S8300 loses contact with its G700 Media Gateway, it will take the media gateway out of service after a time-out period. However, if the G700 loses contact with the S8300, the media gateway first tries to reestablish contact. If it fails after a time-out period, then the G700 tries to connect with another media server in its controller list. When a functional media server is located, the G700 reboots.

If the primary call-processing server goes offline and an LSP is available as a standby unit, it will assume call processing as follows:

- IP phones and media gateways that were previously using the primary media server will try to register with the standby media server for call processing, provided that they have been administered to do so in the controller list (use the `set mgc list` command).

- The standby media server (LSP) will go into license error mode, then start to provide call processing. It cannot preserve any calls set up by the primary media server. IP phone connections can stay up until the call is completed if they are shuffled, but no features are supported on the call.

  NOTE:
  The license error mode will run for up to 6 days and if the problem is not resolved, then it goes into no license mode and call processing halts.

- When the primary media server is available again, it will begin handling call processing. However, those endpoints that are now using the standby media server will continue to do so until the standby unit is taken down.
- If the standby S8300 media server is rebooted, all devices will return to using the primary media server for call-processing service. Any calls in progress on the LSP will be dropped when the reboot occurs (the change back to the primary media server is *not* call preserving).

Using Device Manager to administer G700 Media Gateway components

Device Manager, also known as the Cajun P330 Embedded Web Manager, provides a browser-based graphical user interface (GUI) to assist you with ongoing media gateway administration. For complete information on using Device Manager, see the A*vaya Cajun P330 Manager User Guide*.

Device Manager allows you to:

- View the status of different devices on the network.
- Configure or modify devices including Virtual LAN (VLAN) groupings, port mirroring, and Simple Network Management Protocol (SNMP) traps.

Accessing Device Manager from a web browser

To access the P330 Device Manager from a web browser:

1. Open a compatible web browser on your computer. Microsoft Internet Explorer 5.0 (or higher) and Netscape Navigator 4.7 and 6.2 are supported. The Java Plug-in 1.2.2 or 1.3.1 is required.
2. In the Address (or Location) field of your browser, type the IP address of the Cajun stack.

3. Log in as prompted.
   - The default user name is: `root`
   - The default password for read-write access is: `root`

   NOTE:
   You should change the default passwords after you log in, if they have not already been changed. The passwords apply to all logins for these devices, whether accessed through Device Manager or a CLI.

4. The Welcome Page is displayed. Proceed with Cajun and media gateway device administration.

Accessing Device Manager through MultiService Network Manager

You can alternatively launch the P330 Device Manager from the Avaya MultiService (formerly CajunView) Network Manager. This optional product is a complete Network Management System (NMS). Services include:

- viewing all network devices by type, subnet, or customized groupings
- logging and viewing SNMP traps and events
- launching and managing other applications including ASA

Contact your Avaya representative to obtain a copy of this program if desired.

Command line interface administration

Instead of using Device Manager, you could telnet into the Media Gateway Processor (MGP), or use the serial interface on the front of the chassis to establish a direct connection to the Cajun stack processor.

- Command line interface (CLI) access procedures are covered in *Welcome to the Avaya S8300 Media Server and Avaya G700 Media Gateway.*
- For a list of CLI commands, see the *CLI Reference*.

S8300 Media Server administration

The S8300 Media Server contains the Avaya MultiVantage call-processing software, and controls its operation over the corporate network. Some of the primary functions controlled by the S8300 Media Server are:

- Backing up and restoring call processing, server, and security data.
- Checking server and process status.
- Administering network features for the S8300 media server such as SNMP service, enabling or disabling the modem (if used), enabling FTP services, and installing license and authentication files.
- Installing new software and reconfiguring the server as needed.
- Performing routine diagnostics and troubleshooting such as viewing alarms and system logs, and running tests if needed.

Media Server Web Interface and online help

S8300 Media Server tasks are performed using the Media Server Web Interface. This browser-based server administration tool is an easy-to-use graphical user interface for performing the server administration tasks listed above. It contains an extensive help system that describes all web screens and media server procedures. This section covers highlights of media server administration.

Accessing the Media Server Web Interface

The S8300 Media Server Web Interface can be accessed through the corporate LAN connection to the G700 Media Gateway, or through the Services Ethernet interface on the front of the S8300 Media Server using a crossover cable. Details on how to configure a laptop for a direct connection are in the online help and in *Welcome to the Avaya S8300 Media Server and Avaya G700 Media Gateway.*

To access the media server, you must log in as follows:

1. Open a compatible Internet browser on your computer. Currently only Internet Explorer 5.x (5.5 with Service Pack 2 is recommended) and Netscape 4.7x are supported.
2. In the Address (or Location) field of your browser, type the IP address or name of the Avaya media server and press Enter.
   - *LAN access by IP address.* If you are logging into the administrative interface over the corporate local area network, you can type the media server's unique IP address in standard dotted-decimal notation, such as http://192.152.254.201.

- *LAN access by server name.* If the server has already been configured and if the corporate LAN includes a domain name service (DNS) server that has been administered with the servers' names, you can type the server's name into the address field instead of the IP address. Server names vary depending on local administration (such as http://media-server1.mycompany.com).
- *Laptop access by IP address.* If you are logging in to the services Ethernet interface from a directly connected laptop, the IP address on the server is always `192.11.13.6`. New servers that have not yet been configured can only be accessed in this way. Server-name login is *not* available through the services interface because this connection is a closed (private) network with no DNS.

NOTE:
The S8300 media server's name and IP address are specified during initial media server configuration.

3. *If your browser does not have a valid security certificate,* you will see a warning screen and instructions to load the security certificate.
   - If you are certain your connection is secure, accept the server security certificate to access the Login screen.
   - If you plan to use this computer and browser to access this or other Avaya media servers again, click the main menu link to Install Avaya Root Certificate after you log in.
4. The Login screen appears.
   - In the Username field, type your user name (login ID), such as **cust**.
   - Click the Login button or press Enter.

NOTE:
User names and passwords are case sensitive. Enter the login ID and confirmation information in upper- or lowercase as required.

5. Enter your login confirmation information as prompted:
   - *Password prompt.* Type your password in the Password field, and click Login or press Enter again.
   - *ASG challenge.* If your login is Access Security Gateway (ASG) protected, you will see a challenge screen. Enter the correct response and click Login or press Enter.

6. The server will match your login information against its authentication tables. If the information you entered is recognized as a valid login, the screen displays a two-frame window, with the main menu in the left panel and a usage-agreement notice in the right window.
   - If the main menu in the left panel has only about 10 links, you have accessed the server from a login ID that has restricted permissions. You can adjust the login service levels using the Login Administration screen in ASA or SAT.
   - If the server cannot recognize your user name or password, you receive an authentication failure message. Return to step 5. If you fail to enter the user name and login confirmation correctly 4 times within a few minutes, the Login screen will block further attempts to log in for a while.
7. When you successfully log in to the server, check the top of the left panel. The Avaya media server you are logged into is identified by name and server number. The S8300 media server number is always 1.

Web interface main menu

The tasks you can perform are shown by a list of links in the left panel of the main web interface window. Table 8 lists the links that are available on the main menu.

For help with any of these tasks, click the Help button on the main menu. Alternatively, if you have already clicked on a link, click the About This Screen button on the web interface page to go directly to the help for that specific screen.

Table 8. S8300 Media Server web interface main menu links

This Server: [1] hostname	
Help	
Log Off	
Telephony Administration	**SNMP**
-- Start ASA	-- Configure Trap Destinations
-- Download ASA	-- Start/Stop Master Agent
Utility Software	**Diagnostics**
-- Download VMON	-- View Restart Causes
Data Backup/Restore	-- View System Logs
-- Backup Now	-- Execute Ping

Continued on next page

Table 8. S8300 Media Server web interface main menu links (Continued)

-- Schedule Backup	-- Execute Traceroute
-- View Backup Log	-- Execute Netstat
-- View/Restore Data	-- Test Modem
Alarms and Notification	**Miscellaneous**
-- View Current Alarms	-- Enable/Disable Ethernet Switch Ports
Server	-- Upload Files to Server (via browser)
-- View Summary Status	-- Download Files to Server (from web)
-- View Process Status	**Server Configuration and Upgrades**
-- Shutdown This Server	-- View Software Version
-- Set Server Time/Timezone	-- Configure Server
Security	-- Install New Software Release
-- Enable/Disable Modem	-- Make Server Upgrade Permanent
-- Enable/Disable Anonymous FTP	-- View Partition Status
-- View License Status	-- Reset to Defaults
-- Install License	
-- Install Avaya Authentication	-- View Serial Number(s)
-- Install Avaya Root Certificate	

Media server web interface tasks

Key tasks that administrators typically perform on the S8300 media server are summarized in this section. See the online help for complete information.

Backup and restore data

One of the most important tasks is to set up a schedule to routinely back up system data to a safe location. Because the S8300 Media Server does not include a media storage device for backup purposes, the data must be emailed or transferred using FTP to another server or computer for backup.

The web interface allows you to back up call-processing data (MultiVantage translations), server system data, and security files. It is recommended that you encrypt the backup files to keep this sensitive information secure.

Upgrade software and configuration

Occasionally you may need to install new software on the media server. The new software (and a new license file if required) must be copied to the media server using one of the methods listed in ''Copy files to the server''. The software installation process uses a wizard on the web to guide you through the process.

You may also need to update your server configuration from time to time, and reverify it after a software upgrade. IP addresses for the server and its gateway to the corporate LAN, or for the optional UPS unit, DNS servers, and modem, are specified using the configuration wizard. The wizard also allows you to specify static network routes for sending information over the network, and to update or change the method by which the S8300 media server keeps time.

Copy files to the server

Files must be copied to the media server from another computer or server in the network, or uploaded from a directly connected laptop computer. Files that may be copied to the server include license and authentication files, system announcements, and files for software upgrades. Files may be copied to the server using one of the following methods:

- Upload Files to Server (via browser) link to upload one or more files from your computer to the server's FTP directory using HTTP protocol.
- Download Files to Server (from web) link to copy files to the server from another server on the network; it works like the Upload Files screen.
- Transfer files from another computer or server accessible from the corporate network using FTP or Trivial FTP (TFTP). Files *must* be transferred in binary mode. Either a GUI or CLI FTP program can be used, depending on what is available on your computer.

SNMP administration

You can set up Simple Network Management Protocol (SNMP) services on the server to receive traps or informs (alarms) from devices that report to the S8300 media server, provide a means for a corporate network management system (NMS) to monitor the server, and send alarm notifications to a services agency, to a corporate NMS, or both.

To activate SNMP alarm notification for S8300 devices, use the Configure Trap Destinations screen to set up SNMP destinations in the corporate NMS. SNMP traps for other devices on the network can be administered using Device Manager; see ‘‘Using Device Manager to administer G700 Media Gateway components’’.

SNMP alarms are different from server hardware- or software-generated Operations Support System (OSS) alarms that are recorded in the server logs, and may be reported to Avaya's Initialization and Administration System (INADS) or another services support agency over the server's optional modem interface or through SNMP notifications. Either method, both, or no alarm-reporting method may be used at a given site.

Downloadable programs

In addition to housing the call-processing software, the S8300 Media Server may be used to download the following application software. A downloadable version of these packages may be accessed through the S8300 Media Server Web Interface. This software must be installed on a computer that is running a compatible Microsoft Windows system as noted.

- Avaya Site Administration is the typical way to do telephony administration. It runs on most Windows systems and is available in several languages. See ‘‘Call-processing administration’’ for details.
- Avaya VoIP Monitoring Manager (or VMON program) is used to monitor network quality of service for Avaya IP endpoints using a graphical user interface. Currently it runs only on Windows 2000 systems.

  NOTE:
  The VMON program may not be available on all S8300 Media Servers. It is normally offered as part of the Avaya VisAbility Management Suite. Contact your Avaya representative if you are interested in obtaining this program.

Call-processing administration

The telephony features of the S8300 Media Server are administered using the same commands and procedures as an S8700 Media Server or a legacy DEFINITY Enterprise Communications System. This section summarizes changes that apply to the S8300 Media Server and G700 Media Gateway.

Overview

Avaya MultiVantage software resides on the S8300 Media Server. It can be accessed through either:

- Avaya Site Administration (ASA) features a graphical user interface (GUI) that provides access to SAT commands as well as wizards that provide simplified administration for frequently used features. You can perform most of your day-to-day administration tasks from this interface such as adding or removing users and telephony devices. You can also schedule tasks to run at a non-peak usage time.

  ASA must be installed on a computer running a compatible Microsoft Windows operating system such as Windows 95, 98, NT 4.0, Millennium Edition, and Windows 2000. Once installed, it can be launched from a desktop icon, from the P330 Device Manager, or through a link in the S8300 Media Server Web Interface.

- The System Access Terminal (SAT) program uses a Command Line Interface (CLI) interface for telephony administration. SAT is available through the ASA package or a telnet session to an Avaya media server that contains the Avaya MultiVantage software.

Security Considerations

Security for administration of the G700 Media Gateway is the same as for traditional MultiVantage levels. This means that administration login passwords are passed in plain text with no encryption. Exceptions to this no-encryption policy include:

- The Access Security Gateway (ASG) program that is installed on all Avaya media servers.
- An encrypted web interface to the S8300 media server (refer to the security certificate information in the media server online help)
- Optional encryption for data backups (see ''Backup and restore data'').
- Support for RADIUS authentication for media gateways and Cajun stack elements using the P330 Device Manager. See ''Using Device Manager to administer G700 Media Gateway components''.

Command syntax changes for media modules

The syntax for using the SAT commands for a G700 Media Gateway or S8300 Media Server has changed. In a traditional DEFINITY system, ports are identified by the cabinet number, carrier, slot, and port. For example: 02A0704

Because this numbering convention does not make sense for media modules, a new convention was developed. The numbering convention for the media modules uses the same seven-character field as does a traditional system, but the fields represent the media gateway number, medial module slot (V1 to V4), and port number (00 to 99 are supported; the actual number of ports that can be specified depends on the type of media module).

Example: 001V205

In this example, the 001 represents the media gateway number, the V2 represents the slot number (possibly V1 through V4), and 05 is the port number.

Screen and command change summary

The following screens have been added or changed to accommodate G700 Media Gateways, S8300 Media Servers, and other media modules. Changes relevant to an S8300 or G700 configuration are summarized in this section.

Media Gateway changes

New commands and screens for media gateways include:

- The `media-gateway` administration screen is used to administer G700 Media Gateways and their media modules. Information is similar to the `list media-gateway` screen (next item), but also includes MAC address, network region, location and site data. See Chapter 18, ''Screen reference'' for details.
- The `list media-gateway ['print' or 'schedule']` command shows the list of currently administered gateways. Information includes the media gateway number, name, serial number, IP address, and whether or not this media gateway is currently registered with the call controller. The IP address field is blank until the media gateway registers once, then remains populated.
- The `list configuration media-gateway x` command allows you to list all the assigned ports on the media modules for the G700 Media Gateway specified by its number (`x`).

System-Parameters Customer-Options screen

This screen has been updated to indicate the presence of an S8300 Media Server as a primary or standby call processor as listed below. See Chapter 18, ''Screen reference''for a complete description of this screen.

- The Optional Features section contains a new Local Spare Processor (LSP) field. If it displays a `y` (yes), this S8300 Media Server is configured to provide standby call processing in case the primary media server is unavailable. See ''Local Survivable Processor configuration'' for details. This display-only field can be reset only by the license file.
- Two additional fields in the Optional Features section indicate if the primary call-processing controller is an S8300 Media Server. If traditional port networking is disabled and Processor Ethernet is enabled, an S8300 Media Server is controlling telecommunications.
  - Port Network Support: set to **n** indicates that traditional port networking is disabled. An S8300 is the primary call controller.
  - Processor Ethernet: set to **y** indicates the presence of an S8300 Media Server.

Quality of Service Monitoring

Several screens changes allow you to monitor Quality of Service (QoS) on an S8300 Media Server and G700 Media Gateway configuration. The media gateway can send data to a real-time control protocol (RTCP) server, which in turn monitors the network region's performance. Screens include:

- A new `RTCP Monitor Server` section on the System-Parameters IP-Options screen allows you to enter a single default IP address, server port, and RTCP report period that can be utilized by all administered network regions. This means you do not have to re-enter the IP address each time you access the `ip-network-region` screen.
- The `ip-network-region` screen must also be administered for QoS monitoring (see *Administration for Network Conectivity for Avaya MultiVantage™ Software* for details). The defaults are set so that no administration is needed in the simple, generic case. For situations that require customization, this screen is administered on a per IP network regional basis. Items to customize include:
  - Enabling or disabling of RTCP monitoring
  - Modifications to the report flow rate
  - Changes to the server IP address and server port

- The `list ip-network-region qos` and `list ip-network-region monitor` commands list quality of service and monitor server parameters from the `ip-network-region` screen as follows:
  - `qos` displays VoIP media and call control (and their 802.1p priority values), BBE DiffServ PHB values, RSVP profile and refresh rate.
  - `monitor` displays RTCP monitor server IP address, port number, report flowrate, codec set, and UDP port range parameters.

Media Gateway serviceability commands

Additional commands related to media gateways appear in *Maintenance for S8700 Media Server for Multiconnect Configurations* and *Maintenance for S8700 Media Server for IP-connect Configurations*.These include:

- The `status media-gateways` command provides an alarm summary, busyout summary, and link summary of all configured media gateways.
- Several commands have been modified to support the media gateway port identification format described in ‘‘Command syntax changes for media modules’’. These include:
  - Message Sequence Trace `(mst)`
  - `display errors`
  - `display alarms`

Changed and unused commands

The ‘‘Modified commands’’ section summarizes SAT commands that have been changed to support S8300 Media Servers, G700 Media Gateways, and additional media modules. Commands that are no longer used for this configuration appear in the ‘‘Unused (disabled) commands’’ section.

Modified commands

Some of the SAT commands have been modified to support G700 Media Gateway module port locations. These commands are shown in Table 9 and Table 10. Commands that are unaffected by the media gateway configuration are not shown.

Table 9. Commands Supported by Media Gateway Ports/Modules, 1

add	change	display	remove	list	status
access-endpoint	access-endpoint	access-endpoint	access-endpoint	access-endpoint	access-endpoint
attendant	attendant	attendant	attendant	configuration (all) (board) (ds1) (stations) (trunks)	attendant
ds1	console-parameters	console-parameters	ds1	eda-external-device-alrm	pri-endpoint
eda-external-device-alrm	ds1	ds1	eda-external-device-alrm	hunt-group	signaling-group
hunt-group	eda-external-device-alrm	eda-external-device-alrm	hunt-group	marked-ports	station
personal-CO-line	hunt-group	hunt-group	personal-CO-line	members (trunk-group)	trunk
pri-endpoint	integrated-announcement-board-location	music-sources	pri-endpoint	personal-CO-line	
signaling-group	music-sources	personal-CO-line	signalling-group	pri-endpoint	
station	personal-CO-line	port	station	signaling-group	
trunk-group	pri-endpoint	pri-endpoint	trunk-group	station	
	signaling-group	signaling-group		testcalls	
	station	station		test-schedule	
	system-parameters (features)	system-parameters (features)		trunk-group	
	tenant	tenant			
	trunk-group	trunk-group			

Table 10. Commands Supported by Media Gateway Ports/Modules, 2

busyout/ release	clear	mark	monitor	ping/ traceroute	reset	test	duplicate
access-endpoint	port	port	trunk	node-name	board	access-endpoint	access-endpoint
board	measure-ment ds1)					alarms	station
port						analog-testcall	
station						board	
trunk						ds1-loop	
						eda-external -device-alrm	
						port	
						signaling-group	
						station	
						trunk	

Unused (disabled) commands

The commands listed in Table 11, Table 12, and Table 13 are *not* supported on an S8300 Media Server with a G700 Media Gateway. These commands *not* accessible through ASA or SAT for this configuration.

Table 11. Disabled commands I

add	change	display	remove	list
atm	atm	atm	atm	atm
cabinet	cabinet	cabinet	cabinet	cabinet
data-module	circuit-packs	circuit-packs	data-module	configuration (atm) (carrier) (port-network) (control)
fiber-link	data-module	data-module	fiber-link	data-module
ip-route	ethernet-options	ethernet-options	ip-route	fiber-link
ipserver-interface	fiber-link	failed-ip-network-region	ipserver-interface	ip-route
modem-pool	ip-route	fiber-link	modem-pool	ipserver-interface
pgate	ipserver-interface	ip-route	pgate	isdnpri-testcall
	modem-pool	ipserver-interface		measurement (atm) (clan) (modem pool)
	paging (loudspeaker)	modem-pool		modem-pool
	pgate	paging (loudspeaker)		pgate
	synchronization *(available from the MG CLI only)*	pgate		testcalls
	system-parameters (ipserver-interface)	synchronization		test-schedule
		system-parameters (ipserver-interface)		

Table 12. Disabled commands II

backup	clear	duplicate	enable/disable	format	monitor	recycle	upload
disk	interface	data-module	synchronization-switch	card-mem	system (conn)	carrier	fw-log
	isdnpri-testcall			disk			
	pgate-port						

Table 13. Disabled commands III

status	set	reset	busyout/release	test
atm	ipserver-interface	disk	atm	analog-testcall
cabinet	pnc	fiber-link	data-module	card-mem
card-mem	switch-node-clock	host-adapter	disk	customer-alarm
clan-ip	synchronization	interface	fiber-link	data-module
clan-port	tdm	ipserver-interface	host-adapter	disk
data-module	tone-clock	maintenance	ipserver-interface	ds1-facility
firmware download		packet-interface	modem-pool	environment
interface		pnc	pnc-standby	failed-ip-network-region
ip-board		port-network	tdm	fiber-link
ip-network-region		tone-clock	tone-clock	hardware group
isdnpri-testcall				host-adapter
packet-interface				interface

Continued on next page

Table 13. Disabled commands III (Continued)

status	set	reset	busyout/release	test
pgate-port				isdnpri-testcall
pnc				led (switch-node)
pnc-network				maintenance
switch-node				mass-storage
synchronization				modem-pool
system				network-control
val-ip				packet-interface
				pkt
				synchronization
				tdm
				tone-clock

The following utility commands return an error message from the CLI. However, they are available through the media server web interface:

- `netstat ip-route`
- `ping node-name` and `ping ip-address`
- `trace-route node-name` and `trace-route ip-address`

Screen reference 18

AAR and ARS Digit Analysis Table

Your switch compares dialed S8100 Media Server with CMC1 Media Gateway numbers with the dialed strings on this table and determines the route pattern for the number.

```
change aar analysis                                              Page 1 of X

                              AAR DIGIT ANALYSIS TABLE
                                    Location:  ___      Percent Full:  ___

     Dialed          Total    Route     Call    Node   ANI
     String         Min Max  Pattern    Type    Num    Reqd
 _________________  __  __    _____     ____    ___     n
 _________________  __  __    _____     ____    ___     n
 _________________  __  __    _____     ____    ___     n
 _________________  __  __    _____     ____    ___     n
 _________________  __  __    _____     ____    ___     n
 _________________  __  __    _____     ____    ___     n
 _________________  __  __    _____     ____    ___     n
 _________________  __  __    _____     ____    ___     n
 _________________  __  __    _____     ____    ___     n
 _________________  __  __    _____     ____    ___     n
```

Screen 28. AAR Digit Analysis Table

```
change ars analysis                                               Page 1 of X

                                  ARS DIGIT ANALYSIS TABLE
                                        Location: ___      Percent Full: ___

     Dialed            Total     Route     Call    Node   ANI
     String           Min Max   Pattern    Type    Num    Reqd
  _________________   __  __     _____     ____    ___     n
  _________________   __  __     _____     ____    ___     n
  _________________   __  __     _____     ____    ___     n
  _________________   __  __     _____     ____    ___     n
  _________________   __  __     _____     ____    ___     n
  _________________   __  __     _____     ____    ___     n
  _________________   __  __     _____     ____    ___     n
  _________________   __  __     _____     ____    ___     n
  _________________   __  __     _____     ____    ___     n
  _________________   __  __     _____     ____    ___     n
```

Screen 29. ARS Digit Analysis Table

Location (for the ARS Digit Analysis Table)

This is a display-only field on the ARS Digit Analysis Table.

Valid display	Usage
1 to **44**	Defines the switch location that uses this ARS Digit Analysis Table.
all	Indicates that this ARS Digit Analysis Table is the default for all port network (cabinet) locations. Appears only if the Multiple Locations field is **n** on System Parameters Customer Options.

Percent Full

Displays the percentage (**0 to 100**) of the system's memory resources that have been used by AAR/ARS. If the figure is close to 100%, you can free-up memory resources.

Dialed String

User-dialed numbers are matched to the dialed string entry that most closely matches the dialed number. For example, if a user dials 297-1234 and the AAR or ARS Digit Analysis Table has dialed string entries of 297-1 and 297-123, the match is on the 297-123 entry.

An exact match is made on a user-dialed number and dialed string entries with wildcard characters and an equal number of digits. For example, if a user dials 424, and there is a 424 entry and an X24 entry, the match is on the 424 entry.

Valid entries	Usage
0 to **9**	Enter up to 18 digits that the switch analyzes.
***, x, X**	wildcard characters

Min

Valid entries	Usage
Between **1** and Max	Enter the minimum number of user-dialed digits the system collects to match to the dialed string.

Max

Valid entries	Usage
Between Min and **28**	Enter the maximum number of user-dialed digits the system collects to match to the dialed string.

Route Pattern

Enter the route number you want the switch to use for this dialed string.

Valid entries	Usage
p1 to **p2000**	Specifies the route index number established on the Partition Routing Table
1 to **640**	Specifies the route pattern used to route the call
r1 to **r32**	Specifies the remote home numbering plan area table. Complete this field if RHNPA translations are required for the corresponding dialed string.
node	Designates node number routing
deny	Blocks the call

Call Type (for AAR only)

Enter the call type associated with each dialed string. Call types indicate numbering requirements on different trunk networks. ISDN Protocols are listed in the table below.

Valid entries	Usage
aar	Regular AAR calls
intl	The Route Index contains public network ISDN trunks that require international type of number encodings.
pubu	The Route Index contains public network ISDN trunks that require unknown type of number encodings.
lev0 to lev2	Specify ISDN Private Numbering Plan (PNP) number formats. (See “ISDN Numbering — Private’’ on page 928 for more information.)

ISDN Protocol

Call Type	Numbering Plan Identifier	Type of Numbering
aar	E.164(1)	national(2)
intl	E.164(1)	international(1)
pubu	E.164(1)	unknown(0)
lev0	PNP(9)	local(4)
lev1	PNP(9)	Regional Level 1(2)
lev2	PNP(9)	Regional Level 2(1)

Call Type (for ARS only)

Valid entries	Usage	China # 1 Call Type
alrt	alerts attendant consoles or other digital phones when an emergency call is placed	normal
emer	emergency call	normal
fnpa	10-digit North American Numbering Plan (NANP) call (11 digits with Prefix Digit “1”)	attendant
hnpa	7-digit NANP call	normal
intl	public-network international number	toll-auto
iop	international operator	attendant
locl	public-network local number	normal

Valid entries	Usage	China # 1 Call Type
lpvt	local private	normal
natl	non-NANP	normal
npvt	national private	normal
nsvc	national service	normal
op	operator	attendant
pubu	public-network number (E.164)-unknown	normal
svcl	national(2)	toll-auto
svct	national(2)	normal

Node Number

Valid entries	Usage
1 to **999**	Enter the number of the destination node in a private network if you are using node number routing or DCS. If you complete this field, leave the Route Index field blank.

ANI Reqd

Valid entries	Usage
y/n	Enter **y** if ANI is required on incoming R2-MFC or Russian MF ANI calls. This field applies only if the Request Incoming ANI (non-AAR/ARS) field on the Multifrequency-Signaling-Related System Parameters screen is **n**.
r	Allowed only if “Allow ANI Restriction on AAR/ARS is **y** on Feature Related System Parameters” screen. Use to drop a call on a Russian Shuttle trunk or Russian Rotary trunk if the ANI request fails. Other types of trunks treat **r** as **y**.

AAR and ARS Digit Conversion Table

Your system uses the AAR or ARS Digit Conversion Table to change a dialed number for more efficient routing. Digits may be inserted or deleted from the dialed number. For instance, you can tell the switch to delete a 1 and an area code on calls to one of your locations, and avoid long-distance charges by routing the call over your private network.

```
change aar conversion                                               Page 1 of 2

                      AAR DIGIT CONVERSION TABLE
                                                   Percent Full: ___


  Matching Pattern     Min   Max   Del   Replacement String   Net  Conv ANI Req
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
```

Screen 30. AAR Digit Conversion Table

```
change aar conversion                                               Page 1 of 2

                      ARS DIGIT CONVERSION TABLE

                          Location:  1                  Percent Full: ___
  Matching Pattern     Min   Max   Del   Replacement String   Net  Conv ANI Req
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
  __________________   __    __    __    __________________   ___   _        _
```

Screen 31. ARS Digit Conversion Table

NOTE:
When you access the screen with **display** or **change**, the entries are sorted in the order of the matching pattern. Digits appear before characters.

Location (for ARS only)

This is a display-only field. Values other than “all” appear only if the Multiple Locations field on the System Parameters Customer Options is **y**.

Valid display	Usage
1 to **44**	Defines the switch location for this ARS Digit Conversion Table.
all	Indicates that this ARS Digit Conversion Table is the default for all port network (cabinet) locations.

Percent Full

Displays the percentage (**0 to 100**) of the system’s memory resources that have been used by AAR/ARS. If the figure is close to 100%, you can free-up memory resources.

Matching Pattern

Valid entries	Usage
0 to **9** (1 to 18 digits)	Enter the number you want the switch to match to dialed numbers. If a Prefix Digit 1 is required for 10-digit direct distance dialing (DDD) numbers, be sure the matching pattern begins with a 1.
***, x, X**	wildcard characters

Min

Valid entries	Usage
1 to **24**	Enter the minimum number of user-dialed digits the system collects to match to this Matching Pattern.

Max

Valid entries	Usage
Min to **28**	Enter the maximum number of user-dialed digits the system collects to match to this Matching Pattern.

Del

Valid entries	Usage
0 to Max	Enter the number of digits you want the system to delete from the beginning of the dialed string.

Replacement String

Valid entries	Usage
0 to **9** (1 to 18 digits) *****	Enter the digits that replace the deleted portion of the dialed number. Leave this field blank to simply delete the digits.
#	Use # to indicate end-of-dialing. It must be at the end of the digit-string.

Net

Enter the switch network used to analyze the converted number.

Valid entries	Usage
ext, aar, ars	Analyze the converted digit-string as an extension number, an AAR address, or an ARS address.

Conv

Valid entries	Usage
y/n	Enter **y** to allow additional digit conversion

ANI Req

This field applies only if the Request Incoming ANI (non-AAR/ARS) field on the Multifrequency-Signaling-Related System Parameters screen is **n.**

Valid entries	Usage
y/n	Enter **y** to require ANI on incoming R2-MFC or Russian MF ANI calls.
r	Allowed only if the Allow ANI Restriction on AAR/ARS field is **y** on the Feature Related System Parameters screen. Use to drop a call on a Russian Shuttle trunk or Russian Rotary trunk if the ANI request fails. Other types of trunks treat **r** as **y**.

Abbreviated Dialing List

You use the Abbreviated Dialing List screens to establish system-wide or personal lists for speed dialing.

Enhanced List

The Enhanced Abbreviated Dialing List can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

NOTE:
The Abbreviated Dialing Enhanced List field must be enabled on the System-Parameters Customer-Options screen before you can program an Enhanced List.

You can define only one Enhanced Abbreviated Dialing List in the system. Before you assign numbers to the list, you must define whether you want a 3-digit or 4-digit enhanced list on the Feature-Related System Parameters screen. If you select 3-digit enhanced list, the list can be up to ten separate screens numbered from 0 through 9 that allow you to define up to 1000 numbers. If you select 4-digit enhanced list, the list can include up to 100 separate screens numbered 0 through 99 that allow you to assign up to 10,000 numbers.

If you want your attendants to use abbreviated dialing, you must also administer the Console Parameters screen.

Page 1

```
display abbreviated-dialing enhanced                              Page 1 of 4
                                ABBREVIATED DIALING LIST
                                    Enhanced List
                       Size (multiple of 5): 5                 Privileged? n
    DIAL CODE
      100: ________________________
      101: ________________________
      102: ________________________
      103: ________________________
      104: ________________________
      105: ________________________
```

Screen 32. Abbreviated Dialing Enhanced List

See Abbreviated Dialing for a detailed description of the special characters.

Size (multiple of 5)

The number of dial code list entries you want in this list.

Valid entries	Usage
5 to **100**, in multiples of 5	Up to 100 entries per screen

Privileged

Indicates whether users of this list can dial any number in the list, regardless of the COR of the station from which they dial.

Valid entries	Usage
y/n	Set this field to **n** if you want the system to verify that this station is allowed to dial this number.

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability may be impaired.

Vector Directory Number extension may also be assigned.

Valid entries	Usage
Digits **0** to **9**	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Only use this if network response time is more than 30 seconds

Group List

This screen implements the Abbreviated Dialing Group List. The Group Lists are controlled by the System Administrator. Up to 100 numbers can be entered per group list that can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Field descriptions for page 1

```
display abbreviated-dialing group                                   Page 1 of X
                            ABBREVIATED DIALING LIST
                               Group List: ____
       Size (multiple of 5):   5  Program Ext: ________   Privileged? n
   DIAL CODE
        01: ________________
        02: ________________
        03: ________________
        04: ________________
        05: ________________
```

Screen 33. Abbreviated Dialing Group List

Group List

This is a display-only field when the screen is accessed using an administration command such as **add** or **change**.

Valid entries	Usage
Display-only field	Enter a group number when completing a paper screen.

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

Program Ext

Enter the extension that you want to give permission to program the Group List.

Privileged

Valid entries	Usage
y	If **y** is entered, the calling phone's class of restriction (COR) is never checked and any number in the group list will be dialed.
n	If **n** is entered, the calling phone's COR is checked to determine if the number can be dialed.

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability may be impaired.

Only DIAL CODEs 1 through 5 are displayed initially. If you enter a number greater than 5 in the Size field, the system increases the number of dial codes to the number you specified.

Vector Directory Number extension may also be assigned.

Valid entries	Usage
Digits **0** to **9**	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Only use this if network response time is more than 30 seconds

Personal List

This screen establishes a personal dialing list for phone/data module users. The personal list must first be assigned to the phone by the System Administrator before the phone user can add entries in the list. The lists can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

```
display abbreviated-dialing personal                                  Page 1 of 4
                         ABBREVIATED DIALING LIST

                  Personal List: ________ List Number: ___
                  Size (multiple of 5): 5
DIAL CODE
     01: ________________
     02: ________________
     03: ________________
     04: ________________
     05: ________________
     06: ________________
     07: ________________
     08: ________________
     09: ________________
     00: ________________
```

Screen 34. Abbreviated Dialing Personal List screen

Personal List

Enter the extension of the phone that will use this list.

List Number

Valid entries	Usage
1 to **3**	Identify which of the three personal lists you want to define for the phone.

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability may be impaired.

Only DIAL CODEs 1 through 5 are displayed initially. If you enter a number greater than 5 in the Size field, the system increases the number of dial codes to the number you specified.

Vector Directory Number extension may also be assigned

Valid entries	Usage
Digits **0** to **9**	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Only use this if network response time is more than 30 seconds

System List

This screen implements a system abbreviated dialing list. Only one system list can be assigned and is administered by the System Administrator. The list can be accessed by users to place local, long-distance, and international calls; to activate/deactivate features; or to access remote computer equipment.

Page 1 of 4 of the screen

```
display abbreviated-dialing system                                  Page 1 of X
                              ABBREVIATED DIALING LIST
                                    System List
                     Size (multiple of 5): 5        Privileged? n
         DIAL CODE
            01: ________________
            02: ________________
            03: ________________
            04: ________________
            05: ________________
            06: ________________
            07: ________________
            08: ________________
            09: ________________
            10: ________________
            11: ________________
            12: ________________
            13: ________________
            14: ________________
            15: ________________
```

Screen 35. Abbreviated Dialing System List screen

Size (multiple of 5)

Enter the number of abbreviated dialing numbers you want to assign in multiples of 5, up to 100.

Privileged

Valid entries	Usage
y	Enter **y** if the originating party’s class of restriction (COR) is never checked and any number in the list can be dialed.
n	Enter **n** if the COR is to be checked to determine if the number can be dialed.

DIAL CODE

Enter the number you want the system to dial when users enter this dial code. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability may be impaired.

Only DIAL CODEs 1 through 5 are displayed initially. If you enter a number greater than 5 in the Size field, the system increases the number of dial codes to the number you specified.

Vector Directory Number extension may also be assigned.

Valid entries	Usage
Digits **0** to **9**	Up to 24 characters
***** (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Change to outpulse DTMF digits at the end-to-end rate
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Only use this if network response time is more than 30 seconds

7103A Button List

This screen assigns abbreviated dialing numbers to the 7103A phone buttons. The entries can then be accessed by 7103A phone users to place local, long-distance, and international calls; activate/deactivate features; or to access remote computer equipment. This screen applies only to 7103A fixed feature phones. Only one 7103A abbreviated dialing list can be implemented in the system and it applies to all 7103A fixed feature phones in the system. This list is controlled by the System Administrator.

Pages 1 through 4 of the screen

```
display abbreviated-dialing 7103A-buttons                              Page 1 of 1
                              ABBREVIATED DIALING LIST
                                 7103A Button List

     DIAL CODE (FOR THE 7103A STATION BUTTONS)
      1: ________________________        5. ________________________
      2: ________________________        6. ________________________
      3: ________________________        7. ________________________
      4: ________________________        8. ________________________
```

Screen 36. Abbreviated Dialing List — 7103A Button List

DIAL CODE

Enter the number you want to assign to each dial code (button). Any additions or changes apply to all 7103A fixed feature phones. While the system is waiting, a call progress tone receiver is tied up, and, since there are a limited number of receivers in the system, outgoing calling capability may be impaired.

Vector Directory Number extension may also be assigned.

Valid entries	Usage
Digits **0** to **9**	Up to 24 characters
* (star)	Part of FAC
# (pound)	Part of FAC
~p	Pause 1.5 seconds
~w	Wait for dial tone
~m	Mark
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Only use this if network response time is more than 30 seconds

Access Endpoint

This screen administers Access Endpoints and Wideband Access endpoints.

NOTE:
You can administer Wideband Access Endpoints only if, on the System-Parameters Customer-Options screen, the Wideband Switching field is **y**.

An Access Endpoint is a nonsignaling trunk that neither responds to signaling nor generates signaling. Access Endpoints eliminate the need to dedicate an entire trunk group for the access of a single trunk by providing the capability to assign an extension number to a single trunk.

An Access Endpoint can be specified as the Originator or Destination endpoint of an administered connection.

A Wideband Access Endpoint (WAE) is an endpoint application connected to line-side non-ISDN T1 or E1 facilities and, like Access Endpoints, have no signaling interface with the system. For information on endpoint applications connected to line-side ISDN-PRI facilities, see “PRI Endpoint’’ on page 1024.

The WAE is defined by a starting port (DS0) and a width specifying the number of adjacent nonsignaling DS0s (positioned within a DS1 facility) that make up the endpoint. This width may be between 2 and 31 adjacent DS0s.

NOTE:
Access Endpoints and Wideband Access Endpoints consume the same resources that trunks use. Thus, the sum of Access Endpoints and trunks cannot exceed the maximum number of trunks available in your system configuration.

Field descriptions for page 1

```
add access-endpoint next                                      Page   1 of   1
                              ACCESS ENDPOINT

            Extension: 30001       (Starting) Port:_______
   Communication Type: voice-grade-data       Name:_____________

                  COR: 1                       COS: 1
                   TN: 1                       ITC: restricted
```

Screen 37. Access Endpoint screen

Extension

A display-only field showing the extension number as specified in the command line, or shows the next available extension number if **next** was entered on the command line. This is the extension number assigned to the nonsignaling trunk and used to access the trunk endpoint.

(Starting) Port

Enter seven characters.

Valid entries	Usage
01 through **44** (DEFINITY Server R) or **01** through **03** (DEFINITY Server CSI)	First and second characters are the cabinet number
A through **E**	Third character is the carrier
01 through **20**	Fourth and fifth character are the slot number
01 through **04** (Analog TIE trunks) **01** through **31** (DS1 Interface ports)	Six and seventh characters are the circuit number

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

NOTE:
For Wideband Access Endpoints, analog tie trunks cannot be used and the DS1 Interface circuit pack, Version C or later, must be used.

The DS1 circuit number corresponds to the channel that will carry the data traffic. Channels 1 through 31 (DS1 Interface only) or channels 1 through 24 (DS1 Tie Trunk, DS1 Interface, or DS1 Interface (32) circuit packs) may be used when the DS1 Signaling Type field is **robbed-bit** or **isdn-ext**. For Common Channel or ISDN-PRI signaling, channel use is limited to channels 1 through 30 (DS1 Interface circuit pack only) or channels 1 through 23 (DS1 Interface (32) or DS1 Interface). A channel can be administered as an access endpoint regardless of the DS1 signaling type.

Communication Type

Valid entries	Usage
voice-grade-data	For an analog tie trunk access endpoint.
56k-data **64k-data**	For a DS1 access endpoint enter as appropriate (**64k-data** is not allowed for robbed-bit trunks).
wideband	For a Wideband access endpoint

Name

Enter an name for the endpoint.

Width

Only appears if the Communication Type field is **wideband**. This field cannot be blank.

Valid entries	Usage
2 to **31**	Enter the number of adjacent DS0 ports beginning with the specified Starting Port, that make up the WAE.
6	A width of **6** defines a 384 Kbps WAE.

COR

The COR is administered so that only an administered connection (AC) endpoint can be connected to another AC endpoint.

Valid entries	Usage
0 to **95**	Enter the appropriate class of restriction (COR) number.

COS

The COS is administered (refer to ‘‘Class of Service’’ on page 650) so that the use of the Call Forwarding All Calls feature for access endpoints is prohibited.

Valid entries	Usage
0 through 15	Enter the appropriate COS number.

TN

Valid entries	Usage
1 through **20**	Enter the Tenant Partition number.

ITC (Information Transfer Capability)

This field is used to determine the type of transmission facilities to be used for ISDN calls originating from this endpoint. Displays when the Communication Type field is **56k-data**, **64k-data**, or **Wideband**.

When adding an access endpoint with the ITC administered as unrestricted, its associated port has to be a channel of a DS1 circuit pack with the Zero Code Suppression field administered as B8ZS. If the port is not a channel of a DS1 circuit pack with its Zero Code Suppression field administered as B8ZS, the end validation fails and the screen submission is rejected. The cursor is moved to ITC with the following error message:

```
An unrestricted access endpoint can only be from B8ZS DS1
circuit pack.
```

When adding an access endpoint with the ITC administered as restricted, its associated port can be a channel from a DS1 circuit pack with the Zero Code Suppression field administered as ZCS or B8ZS.

For an existing access endpoint, ITC can only be changed from restricted to unrestricted if its associated port is a channel of a DS1 circuit pack with its Zero Code Suppression field administered as B8ZS. If the port is not a channel of a DS1 circuit pack with its Zero Code Suppression field administered as B8ZS, the end validation fails and the screen submission is rejected. The cursor is moved to ITC with the following error message:

```
An unrestricted access endpoint can use only B8ZS DS1
circuit pack
```

Without this end validation, a user could administer an access endpoint as unrestricted when in fact it is restricted, that is its associated port is a member of a DS1 circuit pack that uses ZCS data transmission.

Valid entries	Usage
unrestricted	When **unrestricted**, only unrestricted transmission facilities (b8zs) will be used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1’s density digital transmission (that is, digital information is sent exactly as is). For Wideband Access Endpoints, enter **unrestricted**.
restricted	When **restricted**, either restricted (zcs-ami) or unrestricted transmission facilities is used to complete the call. A restricted facility is a transmission facility that enforces 1’s density digital transmission (that is, a sequence of eight digital zeros is converted to a sequence of seven zeros and a digital one) via zcs coding on DS1 circuit pack.

Administered Connection

This screen assigns an end-to-end Administered Connection (AC) between two access endpoints or data endpoints. The AC is established automatically by the system whenever the system restarts or the AC is due to be active. Refer to ''Administered Connections'' on page 1338 and ''Access Endpoint'' on page 575 for additional information.

Field descriptions for page 1

```
change administered-connection                              Page 1 of 1
                         ADMINISTERED CONNECTION
  Connection Number:  1                         Enable? y
         Originator:  ________
        Destination:  __________________________________
               Name:  __________________

AUTHORIZED TIME OF DAY

         Continuous? n
                     Sun? n Mon? n Tue? n Wed? n Thu? n  Fri? n  Sat? n
         Start Time:  00:00
           Duration:  000:00

  MISCELLANEOUS PARAMETERS

                 Alarm Type:  warning  Alarm Threshold: 5
                                        Retry Interval: 2
                   Priority:  5       Auto Restoration? y
```

Screen 38. Administered Connection screen

Connection Number

This is a display-only field showing an unassigned AC number when the screen is accessed using an administration command such as **change** or **display**.

Enable

Provides the administered connection.

Valid entries	Usage
y	Indicates an attempt will be made to establish the AC when the AC is due to be active.
n	The AC will not be made or if it is up, it will drop.

Originator

Enter the assigned access endpoint/data module extension.

Data Line circuit pack

- Asynchronous EIA 232C compatible equipment

Digital Line circuit pack connections, including:

- MPDM (700D), MTDM (700B, 700C, 700E), 7400D data module
- 7400A, 7400B, 7400C HSL, 8400B data module
- 7401D phone with 7400B or 8400B data module
- 7403D/7405D/7407D/7410D/7434D phone with DTDM or 7400B or 8400B data module
- 7404D or 7406D phone
- 510D personal terminal
- 515 BCT, 615 BCT, or 715 BCT terminal
- PC/switch connection

ISDN-BRI Line circuit pack connections, including:

- 7500 data module
- 7505D/7506D/7507D phone with ADM

Valid entries	Usage
Assigned access endpoint/data module extension	The endpoint must be local to the switch on which the AC is administered. Nonsignaling DS1 trunk or analog tie trunk.

Destination

Used to route the AC to a desired endpoint. Enter the address of the destination access or data endpoint. This endpoint is the terminating party of the AC and need not be local to the switch on which the AC is assigned. The entry must be consistent with the local switch's dial plan (that is, the first digits are assigned as an extension, feature access code, or trunk access code, or DDD Number). If a local extension is entered, it must be assigned to either an access or data endpoint. Abbreviated Dialing entries may be used in this field.

Valid entries	Usage
Digits **0** to **9**	Up to 36 characters
***** (star)	
# (pound)	
~p	Pause
~w	Wait for dial tone
~m	Mark
~s	Start suppressing display of the digits being outpulsed
~W	Wait indefinitely for dial tone. Only use this if network response time is more than 30 seconds

Name

Valid entries	Usage
Up to 27 alphanumeric characters.	Enter a short identification of the AC.

Authorized Time of Day

Continuous

The connection will be up all the time or re-established if the connection goes down.

Valid entries	Usage
y	Indicates that the AC is continuous (that is, not scheduled to be active at a certain time). If **y** is entered, the seven Start Days and associated Duration fields do not appear.
n	Displays the Start Days fields.

Start Days (Sun through Sat)

These fields indicate only the days on which an attempt will be made to establish the AC and not necessarily the days it is active. A scheduled AC may be active over a number of days, and, in this situation, these fields should be used only to specify the days on which the AC starts and not other days on which the AC may be active. Only appears if the Continuous field is **n**.

Valid entries	Usage
y	Enter **y** in each of the required days of the week fields to indicate that an attempt will be made to establish the AC.
n	Displays the day fields.

Start Time

Only appears if the Continuous field is **n**.

Valid entries	Usage
00:00 through **23:59**	Enter the time of the day when an attempt should begin to establish a scheduled AC. The time is specified in two fields separated by a colon.

Duration

Enter the period of time that the scheduled AC should remain active. This period is specified in two fields separated by a colon. The maximum duration is 167 hours and 59 minutes (that is, 1 minute less than 1 week). Only appears if the Continuous field is **n**.

Valid entries	Usage
000 through **167**	For the hour field.
00 through **59**	For the minute field.

Miscellaneous Parameters

Alarm Type

Enter the type of alarm to be generated if the AC cannot be initially established, or fails and cannot be reestablished, and the number of consecutive failures equals the alarm threshold. All AC alarms and the errors that caused the alarms are recorded in the system's alarm and error log. In addition, a status lamp associated with an attendant console or phone feature button (ac-alarm) may be used to indicate the AC alarm.

Valid entries	Usage
major	Failures that cause critical degradation of service and require immediate attention.
minor	Failures that cause some degradation of service, but do not render a crucial portion of the system inoperable. This condition requires action, but its consequences are not immediate. Problems might be impairing service to a few trunks or stations or interfering with one feature across the entire system.
warning	Failures that cause no significant degradation of service or failures in equipment external to the system. Warning alarms are not reported to the attendant console or INADS.
none	The alarm notification may be disabled for this AC.

Alarm Threshold

Only appears if an entry in the Alarm Type field is other than **none**. Enter the number of times an attempt to establish or reestablish an AC must fail consecutively before an AC alarm generates. (an alarm will be generated after the fourth retry has failed, thus, with the retry interval of 2 minutes, an alarm will be generated approximately 8 minutes after the first failure occurs).

Valid entries	Usage
1 through **10**	An alarm generates on the first failure if this field is **1**.

Retry Interval

Valid entries	Usage
1 through **60**	Enter the number of minutes between attempts to establish or reestablish the AC

Auto Restoration

Valid entries	Usage
y	Enter **y** to indicate an attempt is to be made to reestablish an AC that failed. Auto Restoration is available only for an AC that is established over an ISDN Software Defined Data Network (SDDN) trunk group. A **y** in this field is ignored in all other situations.

Priority

Enter a number that is to be used to determine the order in which ACs are to be established.

Valid entries	Usage
1 through **8**	1 is the highest and 8 the lowest priority

Alias Station

This screen allows you to configure the system so that you can administer new phone types that are not supported by your system software. This screen maps new telephone models to a supported telephone model. This mapping does not guarantee compatibility, but allows nonsupported models to be administered and tracked by their own names.

Some administrators also use this screen to “name” non-telephone device. For example, you know that you can add a modem to your system by simply administering the extension as the standard analog type 2500. But, if you listed your stations, how would you know which extensions are modems? Instead, you could use the Alias screen to create a ‘modem’ alias to type 2500 and enter modem in the Type field for every modem you add to your system. For more information, refer to ‘‘Adding a fax or modem’’ on page 103.

Tip:

When you upgrade a system that uses an alias set type to a new release, the system determines if the aliased type is supported in the new release (is now a native set type). When you review the Alias Station screen, you may see alias types that have become native. If the type is now native, the last character of the aliased set type becomes a “#.”

Field descriptions for page 1

```
change alias station
                                                                Page 1 of 1
                                 ALIAS STATION
                      Alias Set Type     Supported Set Type
                           ______             ______
                           ______             ______
                           ______             ______
                           ______             ______
                           ______             ______
                           ______             ______
                           ______             ______
                           ______             ______
                           ______             ______
                           ______             ______
            '#' indicates previously aliased set type is now native
```

Screen 39. Alias Station screen

Alias Set Type

Enter up to a 5-character name for the nonsupported phone type that you want alias to a similar supported telephone type.

Supported Set Type

Enter a supported phone type that you want to map (or alias) to the alias set type. Valid supported phone types are listed in Table 22 on page 1099.

> **NOTE:**
> Data Communication Protocol (DCP) telephone types must be aliased to DCP telephone types, hybrid types to hybrid types, and analog to analog types.

Alphanumeric Dialing Table

This screen associates alpha-names to dialed digit strings. This allows telephone users to place a *data call* by simply typing the alpha-name. Users need only remember far-end alpha-names instead of the actual digit strings.

The screen consists of paired Alpha-name/Mapped String fields. Entries may be made in any order on the screen. However, before the screen is displayed for changing or reviewing, the entries in the table are sorted alphanumerically by the alpha-name. All entries will be moved to the beginning of the table, leaving all blank entries at the end.

Field descriptions for page 1

```
change alphanumeric-dial-table                                     Page 1 of 2
                              ALPHANUMERIC DIALING TABLE
  XXX of XXX administered
  Alpha-name        Mapped String          |  Alpha-name         Mapped String
                                           |
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
  ____________  ______________________     |  _____________  ______________________
```

Screen 40. Alphanumeric Dial Table screen

Alpha-name

All alpha-names in the table must be unique and cannot be referenced in their own “Mapped String”. The alpha-names can be used in any other “Mapped String” and multiple times in a “Mapped String”.

Valid entries	Usage
From 1 to 8 alphanumeric characters.	The entry must start with an alphabetic character and may not have blank spaces between characters.

Mapped String

Enter from 1 to 24 characters that may contain alphanumeric, readability, delimiters, and/or special characters. The entry is used to generate the final dialing string and can include Facility Access Codes.

NOTE:
A Mapped String may not contain an Alpha-Name whose Mapped String also contains an Alpha-Name.

Valid entries	Usage
Digits **0 - 9**.	Numeric
A through **Z**, **a** through **z**	Alpha (note uppercase entries are mapped to lowercase).
(	Character delimiters used for easy reading of the dial string.
)	Delimiters can consist of both readability and special characters and are used to separate tokens of "alpha-names" or numeric sub-strings. For example, "(205) mt-1234+0000".
/	
-	
+	
%	
"**,**"	
space	
#	Each treated as a numeric.
*****	
^	Treated as a readability character.
~w	Suspend digit outpulsing until a dial tone is detected.
~p	Pause 1.5 seconds. (Used only for outgoing trunk calls. If used internally, the 1.5 second pause is ignored.)
~m	Digits following this character are for end-to-end signaling. Following digits to be outpulsed as tones regardless of type of trunk signaling, pulse, or tone.

Announcements/Audio Sources

Use this screen to assign announcements to circuit packs and port locations.

```
change announcements                                               Page 1 of X
                          ANNOUNCEMENTS/AUDIO SOURCES

 Ann.
 No.  Ext.     Type       COR  TN    Name                     Q QLen Pr Rt Port
 1    ________ _________   1_  1_   ________________________ _ _    _  __ _____
 2    ________ _________   1_  1_   ________________________ _ _    _  __ _____
 3    ________ _________   1_  1_   ________________________ _ _    _  __ _____
 4    ________ _________   1_  1_   ________________________ _ _    _  __ _____
 5    ________ _________   1_  1_   ________________________ _ _    _  __ _____
 6    ________ _________   1_  1_   ________________________ _ _    _  __ _____
 7    ________ _________   1_  1_   ________________________ _ _    _  __ _____
 8    ________ _________   1_  1_   ________________________ _ _    _  __ _____
 9    ________ _________   1_  1_   ________________________ _ _    _  __ _____
 10   ________ _________   1_  1_   ________________________ _ _    _  __ _____
 11   ________ _________   1_  1_   ________________________ _ _    _  __ _____
 12   ________ _________   1_  1_   ________________________ _ _    _  __ _____
 13   ________ _________   1_  1_   ________________________ _ _    _  __ _____
 14   ________ _________   1_  1_   ________________________ _ _    _  __ _____
 15   ________ _________   1_  1_   ________________________ _ _    _  __ _____
 16   ________ _________   1_  1_   ________________________ _ _    _  __ _____
```

Screen 41. Announcements/Audio Sources

Ext

Valid entries	Usage
1 to 7 digits	Enter the extension you assign to this announcement. The following screens can reference this extension: Hunt Group, Coverage Path, Trunk Group (Incoming Destination and Night Destination), Vector, Feature-Related System Parameters (DID/Tie/ISDN Intercept Treatment, Controlled Restriction)

Type

Enter the type of announcement you want to assign to this extension number.

If you enter integrated or integ-rep, complete the Q, Protect, Rate, and Port fields. If you enter analog, ds1-fd, ds1-sa, ds1-ops, or aux-trunk, complete QLen (if Q is **y**) and Port.

Valid entries	Usage
analog	Use to play announcements from an external device for a specific period and hang up when finished. When the device hangs up, the caller hears a click. Connects to switch through analog port. Ringing starts playback.
analog-m	Use for continuous playing music or audio source from an external announcement device.
analog-fd	Use to play announcements from an external device for a specific period and hang up when finished. When the device hangs up, the caller hears a click. Connects to switch through analog port. Ringing starts playback. Sends forward disconnect signal to stop playback.
aux-trunk	auxiliary trunk, use with an external announcement device with a 4-wire “aux” interface.
aux-trk-m	auxiliary trunk, use with continuously playing music or audio sources that do not indicate playback is active.
ds1-fd	Assigned to DS1 ports on circuit packs. Recommended for interfacing for CONVERSANT Line Side T1 ports when CONVERSANT is used. Callers do not hear a click when the device hangs up. Provides a disconnect to stop playback when the announcement is done.
ds1-ops	Callers do not hear a click when the device hangs up.
ds1-sa	Callers do not hear a click when the device hangs up. Provides a disconnect to stop playback when the announcement is done.
integrated	Stored internally on the switch on a special integrated announcement circuit pack. Use for general announcements and VDN of Origin Announcements.
integ-rep	integrated repeating

Refer to Appendix A, “Recorded Announcements,” in *Avaya MultiVantage™ Call Center Software Release 11 Guide to ACD Call Centers*, for more information about the Type field.

COR

Valid entries	Usage
0 – **95**	Enter the class of restriction (COR) you want associated with this announcement

TN

Valid entries	Usage
1 to **20**	Enter the Tenant Partition number, if any.

Name

Valid entries	Usage
up to27-character alpha-numeric filename (no ’.’, ’/’, ’:’, ’*’, ’?’, ’<’, ’>’, '\', ‘.wav’, or blanks in this field for VAL circuit packs only).	Describe the announcement message. For TN2501AP and Avaya MultiVantage™ S8100 Media Server with CMC1 Media Gateway ISSPA announcements, this field is required. The value in this field becomes the filename of the announcement. The “wav.” file extension, which is part of the filename stored on the circuit pack, does not appear. Do not enter “wav.” as part of the filename. Names on a single VAL circuit pack must be unique. The system checks for duplicate filenames on the same VAL circuit pack.

Q (Queue)

Valid entries	Usage
y	Enter **y** to queue calls for the announcement if the announcement type is either integrated, integ-rep or aux-trunk. The caller is always connected to the beginning of the announcement. Enter **y** for ACD and vectoring delay announcements. Call Centers should always use this option.
n	No queue and no barge-in. The caller is always connected to the beginning of the announcement. The announcement does not play if a port is not available.
b	Enter **b** to set up barge-in if the announcement type is either integrated, integ-rep or aux-trunk. You can connect callers to the announcement at any time while it is playing. **NOTE:** The same non-barge-in announcement can be played through more than one port (or all ports) of an integrated circuit pack. The initial request to play an announcement selects an available port on the board that the announcement resides. If there are additional requests to play the announcement while it is playing on another port(s), another port is selected. If all ports are busy, new requests to play announcements go to the integrated announcement system queue (Q field must be **y**). Otherwise, the request to play is denied and processing continues without the caller hearing the announcement. When a port becomes available, all queued calls (up to the platform “calls connected” limit) are connected at the same time to hear the announcement play from the beginning. A barge-in announcement starts playing when first requested and continues playing through a port, repeating until there are no more requests. Call processing simultaneously connects calls to the playing barge-in announcement. Each call remains connected until the requesting feature operation removes the call (for example, wait step times out). Barge-in type announcements never select another port to play the same announcement once it is playing on a specific port.

QLen (Queue Length)

The queue length is the number of calls that can queue for this announcement. The maximum number of queues allowed depends on your system configuration.

The QLen field applies if the Q field is **y** and the type is analog, DS1, or aux-trunk. When the type is integrated or integ-rep, N/A appears in this field. Integrated announcements have a pre-set queue length.

Valid entries	Usage
The maximum number your system allows.	Number of calls that can be queued for this announcement.

Pro

When the type is analog, ds1, or aux-trunk, N/A appears in this field.

Valid entries	Usage
y	Enter **y** to protect the integrated announcement from being deleted or changed by any user. For VAL, after an announcement file resides on the circuit pack (recorded or FTP transfer), you can set this field to **y** to protect the file (read-only)
n	Enter **n** to allow only users with a console permission COS to change or delete an announcement. For VAL the default is **n**, meaning that the announcement file is read-write. Use this value when you initially administer the announcement.

Rate

Enter the recording speed (in 1000 bits/second) for integrated announcements. A different recording speed may be used for each integrated announcement. When the type is analog, ds1, or aux-trunk, N/A appears in this field.

Valid entries	Usage
16	16 kbps (8 minutes and 32 seconds of announcement time per circuit pack or 1 hour and 24 minutes for 10 circuit packs). This rate does not provide a high-quality recording. It is not recommended for customer announcements, but it is adequate for VDN of Origin announcements.
32	32 kbps (4 minutes and 16 seconds of total announcement time).
64	64 kbps (for 2 minutes and 8 seconds of announcement time per circuit pack or 42 minutes for 10 circuit packs). This is the default for VAL.

Port

Valid entries	Usage
circuit pack or port location	Specify the address of the integrated circuit pack or aux trunk, analog line, or DS1 port location for this announcement.

ARS Toll Table

This screen assigns ARS Toll Tables used by Subnet Trunking. Use it to specify whether calls to CO codes listed on the table are toll or non-toll calls. You specify non-toll calls based on the last 2 digits of the distant-end of the trunk group.

```
change ars toll
                              ARS TOLL TABLE: __               Page 1 of 1

                             OFFICE CODES: x00-x99
   00: y   10: y   20: y   30: y  40: y  50: y  60: y  70: y  80: y  90: y
   01: y   11: y   21: y   31: y  41: y  51: y  61: y  71: y  81: y  91: y
   02: y   12: y   22: y   32: y  42: y  52: y  62: y  72: y  82: y  92: y
   03: y   13: y   23: y   33: y  43: y  53: y  63: y  73: y  83: y  93: y
   04: y   14: y   24: y   34: y  44: y  54: y  64: y  74: y  84: y  94: y
   05: y   15: y   25: y   35: y  45: y  55: y  65: y  75: y  85: y  95: y
   06: y   16: y   26: y   36: y  46: y  56: y  66: y  76: y  86: y  96: y
   07: y   17: y   27: y   37: y  47: y  57: y  67: y  77: y  87: y  97: y
   08: y   18: y   28: y   38: y  48: y  58: y  68: y  78: y  88: y  98: y
   09: y   19: y   29: y   39: y  49: y  59: y  69: y  79: y  89: y  99: y
```

Screen 42. ARS Toll Table

ARS TOLL TABLE

Valid entries	Usage
2 through **9**	Identify the number of the ARS Toll Table

OFFICE CODES

Valid entries	Usage
200–**299** through **900**–**999**	Identify the block of numbers on this screen.

00: through 99:

These fields represent the last 2 digits of the codes within the 100-block of numbers. Designate each as a number toll or non-toll call.

Valid entries	Usage
y /**n**	Enter **n** to designate a non-toll CO code.

Attendant Console

This screen assigns an Attendant Console to the system.

Field descriptions for page 1

```
change attendant                                                 Page   1 of   3
                             ATTENDANT CONSOLE 1

         Type: console           Name: 27 character attd cons name
    Extension: 1000             Group: 1               Auto Answer: none
 Console Type: principal           TN: 1               Data Module? n
         Port: 01C1106            COR: 1         Disp Client Redir? n
Security Code:                    COS: 1          Display Language: english
                                                 H.320 Conversion? n

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote           Local Remote             Local Remote
 1: 9                   5:                      9:
 2: 82                  6:                     10:
 3:                     7:                     11:
 4:                     8:                     12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:           5:            9:           13:           17:
 2:           6:           10:           14:           18:
 3:           7:           11:           15:           19:
 4:           8:           12:           16:           20:
```

Screen 43. Attendant Console

Attendant Console x

This is a display-only field when the screen is accessed using an administration command such as **add** or **change**.

Valid entries	Usage
Display-only field	Enter the console number when completing a paper screen.

Type

Valid entries	Usage
console	Indicates the type of attendant console being administered.
302	Use for 302B/C/D or SoftConsole IP attendant.

Extension (Optional)

Enter the extension for the individual attendant console. Individual attendant extensions allow attendants to use features that an attendant group cannot use. For example, extensions can be members of a DDC or UCD group. An individual attendant extension can have its own Class of Restriction and Class of Service.

If you give your attendants an individual extension, users can call the attendant by dialing the extension or you can assign them an abbreviated-dialing button for fast access to the attendant.

Valid entries	Usage
An unassigned extension	If an extension is not assigned, the attendant can only be addressed as a member of the attendant group. If the attendant has a data module, the extension field cannot be blank.

Console Type

Enter this console's intended use. There can only be one night-only or one day/night console in the system unless Tenant Partitioning is administered. Night Service is activated from the principal console or from the one station set per-system that has a nite-serv button.

Valid entries	Usage
principal	Puts the attendant console into night service.
day-only	Will not get night service calls.
night-only	Handles only night service calls.
day/night	Handles day or night service calls.

Port

Enter seven characters.

Valid entries	Usage
01 through **44**	First and second characters are the cabinet number.
01 through **03**	
01	
A through **E**	Third character is the carrier.
01 through **20**	Fourth and fifth characters are the slot number.

Valid entries	Usage
01 through **08** (digital line circuit pack)	Sixth and seventh characters are the circuit number
01 through **16** (2-wire digital circuit pack)	
01 through **24** (2-wire digital circuit pack TN2224)	
ip	SoftConsole IP attendant. You must also have Type set to '302B' and enter a security code. IP is allowed only if the customer option for IP Attendant consoles is enabled.
x	Indicates that there is no hardware associated with the port assignment. An individual attendant extension must be assigned in the Extension field.

Each attendant console requires a port on a digital line circuit pack. For reliability, the attendant consoles should not be assigned to ports on the same digital line circuit pack. For example, if three attendant consoles are to be provided, assign each console to a port on three different digital line circuit pack, if possible. However, if required, all attendant consoles can be assigned to ports on the same digital line circuit pack.

Security Code

Enter the security code required by the SoftConsole IP attendant. The required security code length is determined by Minimum Security Code Length on the Feature-Related System Parameters screen.

Name

Enter the name of this console.

Valid entries	Usage
Up to 27 alphanumeric characters.	Any entry is accepted

Group

Valid entries	Usage
1 to **16**	Enter the Attendant Group number

TN

Valid entries	Usage
1 to **20**	Enter the Tenant Partition number.

COS

Valid entries	Usage
0 through **15**	Enter the class of service (COS) for this attendant console.

COR

Valid entries	Usage
0 through **95**	Enter the class of restriction that reflects the desired restriction.

Auto Answer

Valid entries	Usage
all	Entering **all** in this field indicates an incoming call to an idle attendant will be answered automatically without any action (no button presses required) by the attendant.
acd	Entering **acd** indicates only ACD split/skill calls and direct agent calls can auto answer. Non-ACD calls terminated to an attendant console with Auto Answer set to **acd** ring audibly.
none	Entering **none** causes all calls terminated to this attendant console to receive some sort of audible ringing treatment.

Data Module

Valid entries	Usage
y/n	Enter **y** if the console is to be connected to a data terminal via 7400B or 8400 Data Module. If **y** is entered, complete the Data Module screen (page 4).

Disp Client Redir

This field is administrable only if the Hospitality feature has been enabled on the System-Parameters Customer-Options screen. This field affects the station's display on calls originated from a station with Client Room Class of Service.

Valid entries	Usage
y	When the field is **y**, the redirection information for a call originating from a Client Room and terminating to this station displays. Note: For stations with an "audix" station type, AUDIX Voice Power ports, or ports for any other type of messaging that needs display information, this field must be **y**.
n	When the field is **n**, then for all calls originating from a Client Room (even redirected calls) that terminate to this station, this station's display will not show the redirection information. Only the client name and extension (or room, depending on what is administered on the Hospitality screen) displays.

Display Language

Enter the language in which you want console messages displayed.

Valid entries	Usage
English	
French	
Italian	
Spanish	
user-defined	

H.320 Conversion

Allows H.320 compliant calls made to this phone to be converted to voice-only. Because the system can handle only a limited number of conversion calls, you may need to limit the number of phones with H.320 conversion.

Valid entries	Usage
y/n	Enter **y** for H.320 compliant calls.

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)

Enter the trunk access codes (TACs) for local and remote switches. (There are fields for one local TAC and one remote TAC per button labeled *Local* and *Remote*.) The local TAC (1 to 4 digits) refers to a trunk group or Loudspeaker Paging zone on this switch. Remote TACs are only useful in a private network (including DCS) network. The remote TAC (1 to 3 digits) refers to a trunk group on the remote switch. If a remote TAC is given, then the local TAC must refer to a trunk group that connects directly to the remote switch and is also limited to 1 to 3 digits.

Avaya recommends a DCS trunk be specified as the local TAC between the local and remote switches. If the TAC specified as local between the local and remote switches is not a DCS trunk, the remote trunk cannot be monitored by the local switch.

Valid entries	Usage
1-4 digit number	
*****	May be used as first digit
#	

HUNDREDS SELECT BUTTON ASSIGNMENTS

Enter in the appropriate field (1 through 20), the hundreds group to be associated with a Hundreds Group Select button located on an optional selector console.

Valid entries	Usage
1-3 digit hundreds group (plus prefix, if needed)	Fields 1 through 8 are used when the selector console is a 24A-type console and fields 1 through 20 are used for a 26A-type console. Enter a hundreds group number that represents all but the last two digits of an extension number (for example, the Hundreds Select Button — on the selector console — for extension 3822 would be “38”).

Field descriptions for page 2 (SoftConsole IP Attendant)

```
add attendant                                                   Page 2 of 4
                                   ATTENDANT
IP FEATURE OPTIONS


    IP Emergency Calls: extension              Direct IP-IP Audio Connections? y
Emergency Location Ext:                                   IP Audio Hairpinning? y
```

Screen 44. Attendant Console Data Module (page 2)

IP Emergency calls

Use this field to tell the switch how to handle emergency calls from the SoftConsole IP Attendant. This field appears when the Port field is set to ip on the Attendant Console screen.

CAUTION:

An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It reaches solely the local emergency service in the Public Safety Answering Point area where the telephone system is located. Please be advised that an Avaya IP endpoint does not dial to and connect with local emergency service when dialing from remote locations. You should not use an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not be responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations.

Valid entries	Usage
extension	Enter **extension** to send the attendant extension entered in the Emergency Location Extension field, to the Public Safety Answering Point (PSAP).
block	Enter **block** to prevent the completion of emergency calls. Use this entry for attendants who move around but always have a circuit-switched phone nearby, and for users who are farther away from the switch than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an SoftConsole IP Attendant and the call is blocked, they can dial 911 from a nearby circuit-switched phone instead.
cesid	Enter **cesid** to allow the switch to send the CESID information supplied by the SoftConsole IP Attendant to the PSAP. The end user enters the emergency information into the SoftConsole IP Attendant. Use this entry for SoftConsole IP Attendant that are near enough to the switch that an emergency call routed over the switch's trunk reaches the PSAP that covers the switch. If the switch uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the SoftConsole IP Attendant. If the switch uses CAMA trunks for emergency calls, the end user enters a specific digit string for each SoftConsole IP Attendant location, based on advice from the local emergency response personnel.
option	Enter **option** to allow the attendant to select the option (extension, block, or cesid) that the attendant selected during registration and the SoftConsole IP Attendant reported. Use this entry for extensions that are swapped back and forth between SoftConsole IP Attendants and a phone with a fixed location. The user chooses between **block** and **cesid** on the softphone. A DCP or IP phone in the office automatically selects **extension**.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter to y to save on bandwidth resources and improve sound quality of voice over IP transmissions.

Emergency Location Ext

The Emergency Location Ext field defaults to the attendant's extension. This extension identifies the street address or nearby location when an emergency call is made.

Valid entries	Usage
1-8 digits	Enter the Emergency Location Extension for the SoftConsole IP Attendant

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack on the switch.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack on the switch in IP format, without going through the MultiVantage TDM bus.

Field descriptions for page 3

```
add attendant                                                     Page 3 of 4
                                   ATTENDANT
DATA MODULE
     Data Extension: ____              BCC: 2        ITC: restricted
               Name: _______________   COR: 1_       COS: 1_
                                        TN: 1_
ABBREVIATED DIALING
 List1: ________
SPECIAL DIALING OPTION: default
   DEFAULT DIALING
        Abbreviated Dialing Dial Code (From above list): _
ASSIGNED MEMBER (Station with a data extension button for this data module)
        Ext      Name
     1:
```

Screen 45. Attendant Console Data Module (page 3)

This page displays if the Data Module field on Page 1 is **y**. If this is a non-IP attendant console this is page 2 of the Attendant screen.

Data Extension

Enter the extension number assigned to the data module.

Valid entries	Usage
1- to 5-digit number	Must agree with the system’s Dial Plan

BCC

Only displays when the ISDN-PRI or ISDN-BRI Trunks field is enabled on the System-Parameters Customer-Options screen.

NOTE:
The BCC value is used to determine compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking feature).

ITC

Valid entries	Usage
restricted	
unrestricted	

Name

Enter the name of the user associated with the data module. The name is optional, it can be left blank.

COR

Enter the desired class of restriction (COR) number.

Valid entries	Usage
0–95	

COS

Enter the desired (COS) number to designate allowed features. See ‘‘Class of Service’’ on page 650 for additional information on the allowed features.

Valid entries	Usage
0–15	

TN

Enter the Tenant Partitioning number.

Abbreviated Dialing

List1

Valid entries	Usage
s	System
g	Group. If **g** is entered, a group number is also required.
p	Personal. If **p** is entered, a personal list number also is required.
e	Enhanced

SPECIAL DIALING OPTION

Enter one of three dialing options that are available. This identifies the destination of all calls when this data module originates calls.

Valid entries	Usage
hot-line	
default	

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

Only displays when the Special Dialing Option field is **hot-line**. The associated AD number is dialed when the user goes off-hook on a Data Hot Line call.

Hot Line Service allows single-line telephone users, by simply lifting the handset, to automatically place a call to a preassigned destination (extension, telephone number, or feature access code).

The Hot Line Service destination number is stored in an Abbreviated Dialing List.

A Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension, or any individual extension within a group can be a Hot Line Service destination. Also, any extension within a DDC group, UDC group, or TEG can have Hot Line Service assigned.

Use Hot Line Service when very fast service is required and when you use a telephone only for accessing a certain facility. Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

Valid entries	Usage
A dial code	Within the range of the abbreviated dialing list type

DEFAULT DIALING Abbreviated Dialing Dial Code

The associated AD number is dialed when the user goes off-hook and enters a carriage return following the “DIAL” prompt. The data call originator also can perform data terminal dialing by specifying a dial string that may or may not contain alphanumeric names. Only displays when the Special Dialing Option field is **default**.

Valid entries	Usage
A dial code	Within the range of the abbreviated dialing list type

Ext

This is the extension number of a previously administered user who has an associated Data Extension button and who will share the use of the module.

Name

Contains the name assigned to the above extension number.

Field descriptions for page 4

```
change attendant                                                 Page Y of X
                                 ATTENDANT CONSOLE
FEATURE BUTTON ASSIGNMENTS
  1: split_____                           13: __________
  2: __________                           14: __________
  3: __________                           15: __________
  4: __________                           16: __________
  5: __________                           17: __________
  6: hold _____ *                         18: __________
  7: __________                           19: forced-rel
  8: aux-work      RC:    Grp:            20: __________
  9: __________                           21: __________
 10: __________                           22: __________
 11: __________                           23: night-serv *
 12: __________                           24: pos-busy__ *
```

Screen 46. Attendant Console

If this is a non-IP attendant console this is page 3 of the Attendant screen.

FEATURE BUTTON ASSIGNMENTS

Enter the feature buttons from ‘‘Attendant console feature buttons’’ on page 168 that you want to assign to the attendant console. The fixed buttons that cannot be changed (that is, split and forced release) are shown on the screen. The hold, night-serv, and pos-busy buttons are shown in the system default locations. These buttons can be administered elsewhere on the screen. The following provides descriptions of feature buttons that are unique to the attendant console. Refer to the ‘‘Attendant console feature buttons’’ and to the ‘‘Telephone feature buttons’’ sections for more information about the buttons.

Valid entries	Usage
Audible Tones On/Off	
cw-ringoff	Call waiting ringer off; turns on/off the audible tone for call waiting on attendant console (1 per console).
in-ringoff	Incoming call ringer off; turns on/off the audible tone for incoming call ringer (1 per console).
re-ringoff	Timed reminder ringer off; turns on/off the audible tone for timer reminder ringer (1 per console).
alt-frl	Alternate FRL. Alternate facility restriction level; allows the attendant to activate or deactivate the AFRL feature. When activated, this allows the originating device (lines or trunks) to use an alternate set of the facility restriction levels to originate a call (1 per console).
Attendant Control of Trunk Group Access	
act-tr-grp	Activate trunk group access; allows the attendant to control a trunk group. All calls going to the trunks are routed to the attendant (1 per console).
deact-tr-g	Deactivate trunk group access; allows the attendant to release control of a trunk group (1 per console).
class-rstr	Display Class of Restriction. Used to display the COR associated with a call (1 per console).
em-acc-att	Emergency Access to the Attendant. The associated status lamp is flashed when there is one or more calls on the emergency attendant queue (1 per console).
hold	Hold. When the HOLD button is pressed while the attendant is active on a loop, the party on the loop is put on hold and the “call type” button associated with the loop is lit (1 per console).

Valid entries	Usage
pos-busy	Position Busy. When this button is pushed, the attendant is put into position busy mode, the “Pos Avail” light is turned off, and the light associated with the pos-busy button is lit. Pushing the POS-BUSY button a second time takes the console out of “position busy” mode, turns on the “Pos Avail” light and turns off the light associated with the pos-busy button. If the pos-busy button is administered on a 2-LED button, the top LED flashes when the last attendant goes into “Position Busy” mode. Otherwise, if the button has only one LED, the single LED associated with the pos-busy button flashes (1 per console).
serial-cal	Serial Call. This button allows the attendant-extended calls to return to the same attendant if the trunk remains off-hook (1 per console).
override	Attendant Override. This button enables the attendant to override diversion features such as, Call Forwarding, Call Coverage, and so on (1 per console).
intrusion	Call Offer. Depression of this button allows the attendant to extend a call when the called party is active on another call (1 per console).
dont-split	Don’t Split. This button allows the attendant to not split away a call when dialing (1 per console).
vis	Visually Impaired Attendant Service (**vis**) — This button activates visually impaired service for the attendant. When this service is activated, the attendant can listen to console status or messages by pressing buttons that have been translated as follows: ■ “con-stat” repeats the console status. ■ “display” calls out display contents. ■ “dtgs-stat” calls out the DTGS status. ■ “last-mess” repeats the last message. ■ “last-op” calls out the last operation.

Trunk Group Select — In addition to the 12 Direct Trunk Group Selection (DTGS) Button Assignments on Field descriptions for page 1, up to 12 single lamp DTGS buttons can be administered on this page. The status lamp associated with the feature button is used to monitor the busy/idle status of the trunk. Trunk groups administered on these buttons cannot be controlled using Attendant Control of Trunk Group Select buttons. The single lamp DTGS buttons can be administered as follows:

Valid entries	Usage
local-tgs	Local trunk group select; allows the attendant to access trunk groups on the local switch (combination of 12 local-tgs/remote-tgs per console).
remote-tgs	Remote trunk group select; allows the attendant to access trunk groups on a remote switch (combination of 12 local-tgs/remote-tgs per console).
hundrd-sel	Hundreds group select; in addition to the fixed HGS buttons on Field descriptions for page 1, a user can administer hundreds group select feature buttons on this page. When a feature button is administered as “hundrd-sel,” a subfield appears that must then be administered in the same manner as the fixed HGS button fields (a 1 to 3 digit hundreds group plus prefix, if needed). Administered “hundrd-sel” feature buttons operate in the same manner as fixed HGS buttons. The total number of hundreds group select buttons (fixed and administered) allowed on a console is 20. Thus, if all 20 fixed HGS buttons have been administered, no “hundrd-sel” feature buttons can be administered. Note: If no fixed HGS buttons are administered, 19 “hundrd-sel” feature buttons are available. This is because 5 of the 24 feature buttons must be used for required feature buttons (hold, pos-busy, night-serv, forced-rel, and split)
group-disp	Group Display. Allows the attendant to see a display of extensions currently being tracked on the DXS module.
group-sel	Group Select. Allows the attendant to select a specific group of hundreds by dialing the first 2 or 3 digits of the hundreds group.

Attendant Room Status

Valid entries	Usage
occ-rooms	Occupied rooms; allows the attendant to see which rooms are occupied.
maid-stat	Maid status; allows the attendant to see which rooms are in one of six specified states.
vu-display	VuStats (**vu-display**) — This button allows users with display telephones and attendants to turn on the VuStats display. The limit to the number of VuStats feature buttons you can administer depends on how many feature buttons are available on the attendant console you are administering. The system is designed to allow you to set up a separate VuStats display format for each feature button. Therefore, agents can change the type of measurements on their display by selecting a different VuStats feature button.

- If 12 HGS buttons are assigned on Field descriptions for page 2, it is recommended that the “night,” “pos-busy,” and “hold” buttons be reassigned to locations 20, 21, and 3, respectively. The HGS buttons should then be assigned to the right-most three columns, as required.

```
change attendant                                                    Page Y of X
                                    ATTENDANT CONSOLE
DISPLAY MODULE BUTTON ASSIGNMENTS
  1: normal____                                 5: delete-msg
  2: inspect___                                 6: call-disp_
  3: cov-msg-rt                                 7: date-time_
  4: next______                                 8: timer_____
```

Screen 47. Attendant Console

- **DISPLAY MODULE BUTTON ASSIGNMENTS** — Display-type buttons obtain display functions on the associated alphanumeric display. These buttons are noted as [display button] in the Feature or Function column on the table. Also, several feature buttons can be administered so that their associated status lamps can be used to provide visual indications of the associated feature or function. In some cases, the button itself is not operational. These buttons are noted as [status lamp]. If a Call Cover Msg Rt (cov-msg-rt) button is assigned, a Leave Word Calling Delete Msg (delete-msg) button and a Next (next) button must also be assigned.

Audix-MSA Node Names

```
change node-names audix-msa                                        Page 1 of x

                          AUDIX-MSA NODE NAMES

 Audix Name     IP Address          IP Address
 audixA_            __.__.__.__         __.__.__.__
 audixB_            __.__.__.__         __.__.__.__
 _______            __.__.__.__         __.__.__.__
 _______            __.__.__.__         __.__.__.__
 _______            __.__.__.__         __.__.__.__
 _______            __.__.__.__         __.__.__.__
 _______            __.__.__.__         __.__.__.__
 _______            __.__.__.__         __.__.__.__
```

Screen 48. Audix-MSA Node Names screen

Audix Names

Identifies the name of the AUDIX node.

Valid entries	Usage
1-7 character string	Used as a label for the associated IP address. The node names must be unique on each switch.

IP Address

The IP address associated with the node name. This field can be blank for X.25 connections.

Authorization Code — COR Mapping

You use this screen to assign authorization codes and the class of restriction (COR) that is associated with a given authorization code. Refer to *''Authorization codes'' on page 1363* and *''CLAN and processor CLAN QoS and CIDR support'' on page 1521* for more information on how Authorization Codes work with COR.

To maximize the security of your system:

- Administer authorization codes to the maximum length allowed by the system
- Create random (nonconsecutive) authorization codes
- Change authorization codes at least quarterly
- Deactivate authorization codes immediately if a user leaves the company or changes assignments
- Assign each authorization code the minimum level of calling permissions required

Field descriptions for page 1

```
change authorization-code                                 Page 1 of 1
                       Authorization Code - COR Mapping
      Note:    XX codes administered.  Use “list” to display all codes.
     AC    COR   AC    COR   AC    COR   AC    COR   AC     COR    AC    COR
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
    _______ __  _______ __  _______ __  _______ __   _______ __   _______ __
```

Screen 49. Authorization Code - COR Mapping screen

Number of Codes Administered

Displays the number of Authorization Codes already administered using the Authorization Codes screen. There is a maximum number of authorization codes that you can use. To find out what this number is for your system, type **display capacity**, and page down to find the authorization code information.

AC

Valid entries	Usage
Any combination of between 4 and 13 digits	The number of digits must agree with the number assigned to the Authorization Code Length field on the Feature-Related System Parameters screen. To enhance system security, choose Authorization Codes of 13 random digits.

COR

Valid entries	Usage
0–95	When a user dials the associated authorization code, this is the COR that the telephone or other facility will assume for that call.

Bulletin Board

Use the bulletin board to post and receive information. There are three pages of message space within the bulletin board. The first page has 19 lines, but you can only enter text on lines 11-19. The first 10 lines on page 1 are for high-priority messages from Avaya personnel and are noted with an asterisk (*). The second and third pages each have 20 lines, and you can enter text on any line. The system automatically enters the date the message was posted or last changed to the right of each message line.

You can enter up to 40 characters of text per line. You can also enter one blank line. If you enter more than one blank line, the system consolidates them and displays only one. The system also deletes any blank line if it is line one of any page. You cannot indent text on the bulletin board. The TAB key moves the cursor to the next line.

Field descriptions for page 1

```
change bulletin-board                                          Page 1 of 3

              Message (* indicates high-priority)                         Date
          *Avaya is in the process of                                  03/02/93
          *investigating your trunk lockup problem.                     03/02/93
          *The Bulletin Board will be updated as                        03/02/93
          *we find information.                                         03/02/93
          *  We have identified the problem.                            03/04/93
          *The trunk you added does not provide                         03/04/93
          *disconnect supervision. However, the                         03/04/93
          *trunk group was administered as such.                        03/04/93
          *Please call Pat J. for details.                              03/04/93

          We recently added a new trunk group (14)                      03/02/93
          and have had many of the members getting                      03/02/93
          locked up.                                                    03/02/93
          We see the error - thanks for checking.                       03/05/93
```

Screen 50. Sample Bulletin Board

Lines 1 through 10

These lines are reserved for high priority messages and are noted with an asterisk (*) in the first column on the left. If you have an init or inads login you can enter high-priority information to trigger the high-priority message at login time.

Valid entries	Usage
A through **Z**	Enter any information.
a through **z**	
Blank	
0 through **9**	
!@#$%^&*()_ +-=[]{}|\’˜;:’,”<.>/?	

Lines 11 through 19

These lines can be used by anyone with access.

Valid entries	Usage
A through **Z**	Enter any information.
a through **z**	
Blank	
0 through **9**	
!@#$%^&*()_ +-=[]{}|\’˜;:’,”<.>/?	

Date

This display-only field contains the date the information was entered or last changed.

Field descriptions for pages 2 and 3

Lines 1 through 20

These lines can be used by anyone with access.

Valid entries	Usage
A through **Z**	Enter any information.
a through **z**	
Blank	
0 through **9**	
!@#$%^&*()_ +-=[]{}|\’˜;:’,”<.>/?	

Date

This display only field contains the date the information was entered or last changed.

Call Vector

This screen programs a series of commands that specify how to handle calls directed to a Vector Directory Number (VDN). Refer to the *Avaya MultiVantage™ Call Center Software Release 11 Call Vectoring and Expert Agent Selection (EAS) Guide* for additional information.

Field descriptions for page 1

```
change vector 129                                                 Page 1 of 3
                               CALL VECTOR
  Number: 129             Name: _______________    Multimedia? n
            Attendant Vectoring? n         Meet-me Conf? y           Lock? n
    Basic? y   EAS? n    G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? n
Prompting? y   LAI? n   G3V4 Adv Route? y   CINFO? y   BSR? n      Holidays? n

  01 ____________
  02 ____________
  03 ____________
  04 ____________
  05 ____________
  06 ____________
  07 ____________
  08 ____________
  09 ____________
  10 ____________
  11 ____________
```

Screen 51. Call Vector

Field descriptions for page 2

```
                                                           Page 2 of 3
                               CALL VECTOR

  12 ____________
  13 ____________
  14 ____________
  15 ____________
  16 ____________
  17 ____________
  18 ____________
  19 ____________
  20 ____________
  21 ____________
  22 ____________
```

Screen 52. Call Vector

Field descriptions for page 3

```
                                                    Page 3 of 3
                              CALL VECTOR

  23 ____________
  24 ____________
  25 ____________
  26 ____________
  27 ____________
  28 ____________
  29 ____________
  30 ____________
  31 ____________
  32 ____________
```

Screen 53. Call Vector

Number

A display-only field when the screen is accessed using a **change** or **display** administration command.

Name

Represents the vector name.

Valid entries	Usage
Up to 27 alphanumeric characters.	This is an optional field. If ~r can be used to activate Network Call Redirection if, on the System-Parameters Customer-Options screen, the ISDN Network Call Redirection field is **y**.

Multimedia

Indicates whether the vector should receive early answer treatment for multimedia calls. This only applies if the Multimedia Call Handling field is **y**.

Valid entries	Usage
y/n	If you expect this vector to receive multimedia calls, set this field to **y**. If this value is **y**, the call is considered to be answered at the start of vector processing, and billing for the call starts at that time. Refer to ''Managing multimedia calling'' on page 269 for more information.

Attendant Vectoring

This field appears only if, on the System-Parameters Customer-Options screen, the Attendant Vectoring field is **y**. Attendant Vectoring and Meet-me Conference cannot be enabled at the same time.

Valid entries	Usage
y	Enter **y** so the vector is an attendant vector. Default when, on the System-Parameters Customer-Options screen, the Vectoring (Basic) and Vectoring (Prompting) fields are **y**.
n	Default when, on the System-Parameters Customer-Options screen, the Vectoring (Basic) and Vectoring (Prompting) fields are **n**. No changes allowed to this field.

Meet-me Conf

This field appears only if, on the System-Parameters Customer-Options screen, the Enhanced Conferencing field is **y**. This field designates the VDN as a Meet-me Conference VDN.

Valid entries	Usage
y/n	Enter **y** to enable Meet-me Conference for this vector. If Meet-me Conference is **y**, the Lock field also must be **y**. When the Lock field is **y**, the vector cannot be changed by adjunct vectoring programs such as Visual Vectors. Attendant Vectoring and Meet-me Conference cannot be enabled at the same time.

Lock

This field controls access to the vector from Avaya CentreVu® products.

NOTE:
Always lock vectors that contain secure information (for example, access codes).

Valid entries	Usage
y	You do not want this vector to be accessible to these client programs. Locked vectors can only appear and be administered through the SAT or a terminal emulator. If Meet-me Conference is **y**, the Lock field also must be **y**.
n	Gives CentreVu® CMS and CentreVu® Control Center users the ability to administer this vector from these client programs.

Basic

A display-only field indicating whether, on the System-Parameters Customer-Options screen, the Vectoring (Basic) field is **y**.

EAS

A display-only field indicating whether, on the System-Parameters Customer-Options screen, the Expert Agent Selection (EAS) field is **y**.

NOTE:
When Expert Agent Selection (EAS) field is **y**, the help messages and error messages associated with this screen will reflect a terminology change from “Split” to “Skill”. In addition, the vector commands entered also will be affected by this terminology change (for example, *check backup split* becomes *check backup skill* when EAS is enabled).

G3V4 Enhanced

A display-only field indicating whether you can use G3V4 Enhanced Vector Routing commands and features.

ANI/II-Digits

A display-only field indicating whether you can use ANI and II-Digits Vector Routing Commands. ANI/II-Digits Routing requires that the `G3V4 Enhanced` field be **y**.

ASAI Routing

A display-only field indicating whether, on the System-Parameters Customer-Options screen, the CallVisor Adjunct/Switch Applications Interface (ASAI Link Core Capabilities) field is **y**.

Prompting

A display-only field indicating whether, on the System-Parameters Customer-Options screen, the Vectoring (Prompting) field is **y**.

LAI

A display-only field indicating whether Look-Ahead Interflow is enabled.

G3V4 Adv Route

A display-only field indicating whether you can use the G3V4 Advanced Vector Routing commands.

CINFO

A display-only field indicating whether, on the System-Parameters Customer-Options screen, the Vectoring (CINFO) field is **y**.

BSR

A display-only field indicating that on the System-Parameters Customer-Options screen, the Vectoring (Best Service Routing) field is **y**. Thus, you can use BSR commands and command elements in your vectors. An **n** indicates that the BSR option is not enabled.

Holidays

A display-only field that appears when, on the System-Parameters Customer-Options screen, the Vectoring (Holidays) field is **y**.

01 through XX

Enter vector commands as required (up to the maximum allowed in your configuration). For more information, refer to *Avaya MultiVantage™ Call Center Software Release 11 Call Vectoring and Expert Agent Selection (EAS) Guide*.

Valid entries	Usage
adjunct	Causes a message to be sent to an adjunct requesting routing instructions.
announcement	Provides the caller with a recorded announcement.
busy	Gives the caller a busy signal and causes termination of vector processing.
check	Checks the status of a split (skill) for possible termination of the call to that split (skill).

Valid entries	Usage
collect	Allows the user to enter up to 16 digits from a touch-tone phone, or allows the vector to retrieve Caller Information Forwarding (CINFO) digits from the network.
consider	Defines the resource (split, skill, or location) that is checked as part of a Best Service Routing (BSR) consider series and obtains the data BSR uses to compare resources.
converse-on	Delivers a call to a converse split (skill) and activates a voice response script that is housed within a Voice Response Unit (VRU).
disconnect	Ends treatment of a call and removes the call from the switch. Also allows the optional assignment of an announcement that will play immediately before the disconnect.
goto	Allows conditional or unconditional movement (branching) to a preceding or subsequent step in the vector.
messaging	Allows the caller to leave a message for the specified extension or the active or latest VDN extension.
queue-to	Unconditionally queues a call to a split or skill and assigns a queueing priority level to the call in case all agents are busy.
reply-best	Used only in status poll vectors in multi-site Best Service Routing applications, where it “returns” best data for its location to the primary vector on the origin switch.
route-to	Routes calls either to a destination that is specified by digits collected from the caller or an adjunct (route-to digits), or routes calls to the destination specified by the administered digit string (route-to number).
stop	Halts the processing of any subsequent vector steps.
wait-time	Delays the processing of the next vector step if a specified delay time is included in the command’s syntax. Also provides feedback (in the form of silence, ringback, or music) to the caller while the call advances in queue.

CAMA Numbering Format

This screen administers the Centralized Automatic Message Accounting (CAMA) trunks and provides Caller's Emergency Service Identification (CESID) information to the local community's Enhanced 911 system through the local tandem office.

This screen provides the CESID format by extension number or number blocks. This allows for multiple CESID formats to be sent over multiple CAMA trunk groups allowing for mixed station numbering plans and some limited conversion from non-DID to DID numbers typically required by the Private Switch/Automatic Location Interface (PS/ALI) database.

The default CESID defines the CESID for all extensions that are not defined in the Ext Code field.

There are 446 CESID entries over 15 pages. The first page contains the Default CESID and 26 extensions to CESID entries. The second through fifteenth pages each contain 30 extensions to CESID entries.

Field descriptions for page 1

```
change cama-numbering                                           Page 1 of 15
                         CAMA NUMBERING - E911 FORMAT

System CESID Default: _____________

Ext   Ext                 Total          Ext   Ext                   Total
Len   Code       CESID    Length         Len   Code        CESID     Length
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
 _   _______ _____________  __            _   _______ ________________  _
```

Screen 54. CAMA Numbering Format

System CESID Default

Valid entries	Usage
1 to 16 digits	Enter a default CESID. This number will be sent over the CAMA trunk if the Ext Code field does not have an entry.

Ext Len

Valid entries	Usage
1 to **5**	Enter the number of digits in the extension.

Ext Code

Valid entries	Usage
Up to 7 digits	Enter the leading digits or all of the digits in the extension for the specified CESID. If the extension length is greater than the number of digits in the extension code, the extension code will be interpreted as a block of digits. For example, if the extension length is 4 and the extension code is 11, the CESID will serve extensions 1100 through 1199. The Ext Code 11 is for a DID block. An Ext Code of 126 might point a non-DID block to a nearby DID extension 5241666.

CESID

Valid entries	Usage
1 to 11 digits	Enter the number that will be used to identify the calling terminal within an emergency service system. This field may represent a prefix to an extension or the entire CESID.

Total Length

Valid entries	Usage
1–16	Enter the total number of digits to send.

CDR System Parameters

Use the Call Detail Recording (CDR) System Parameters screen to set parameters for the types of calls you want to record and how to format the information. You can use CDR records to determine call costs, diagnose problems, detect abuse, and optimize your network.

Field descriptions for page 1

```
change system-parameters cdr                                    Page   1 of   1
                            CDR SYSTEM PARAMETERS

 Node Number (Local PBX ID):                       CDR Date Format: month/day
      Primary Output Format: printer       Primary Output Endpoint: CDR1
    Secondary Output Format:
          Use ISDN Layouts? n                          EIA Device Bit Rate: 9600
       Use Enhanced Formats? n      Condition Code 'T' for Redirected Calls? n
Modified Circuit ID Display? n                  Remove # From Called Number? n
                  Record Outgoing Calls Only? y            Intra-switch CDR? n
  Suppress CDR for Ineffective Call Attempts? y          CDR Call Splitting? y
      Disconnect Information in Place of FRL? n    Attendant Call Recording? y
                                                     Interworking Feat-flag? n
 Force Entry of Acct Code for Calls Marked on Toll Analysis Form? n
                                    Calls to Hunt Group - Record: member-ext
Record Called Vector Directory Number Instead of Group or Member? n
         Record Called Agent Login ID Instead of Group or Member? n
     Inc Trk Call Splitting? n
  Record Non-Call-Assoc TSC? n
      Record Call-Assoc TSC? n   Digits to Record for Outgoing Calls: dialed
   Privacy - Digits to Hide: 0               CDR Account Code Length: 4
```

Screen 55. CDR System Parameters screen

Node Number (Local PBX ID)

This field displays the DCS switch node number in a network of switches.

CDR Date Format

Use this field to select the format for the date stamp that begins each new day of call records.

Valid entries	Usage
month/day **day/month**	Choose the format that is most appropriate for your situation. If your company has many different sites, you may need to use the same format as the other locations.

Primary Output Format

Controls the format of the call records sent to the primary output device.

Valid entries	Usage
customized	Use this option if you have special call accounting needs that standard record formats do not accommodate. If you use a customized record format, you need to have call accounting software that is also customized to receive these records. Consult with your call accounting vendor before using this option.
printer	Use **printer** if you are sending the call detail records to a printer rather than to a record collection or call accounting system.
59-char **expanded** **lsu** **lsu-expand** **int-direct** **int-isdn** **int-process** **teleseer** **unformatted**	The remaining formats are standard record formats. The one you use must be compatible with your call accounting software. Verify this through your vendor or the accounting system documentation.

Primary Output Endpoint

This field determines where the switch sends the CDR records, and is required if you specify a Primary Output Format.

Valid entries	Usage
eia	If you use the EIA port to connect the CDR device, enter **eia**. This is not a valid option on DEFINITY Server R systems.
Extension number	This is the extension of the data module (if used) that links the primary output device to the switch.
CDR1, **CDR2**	Use this value if the CDR device is connected over a TCP/IP link, and this link is defined as either CDR1 or CDR2 on the IP Services screen.

Secondary Output Endpoint

Appears when the secondary output format is administered.

Valid entries	Usage
eia	Use this if the secondary output device is connected to the eia port. This is not a valid option on DEFINITY Server R systems.
Extension number	This is the extension of the data module (if used) that links the secondary output device to the switch.
CDR1, **CDR2**	Use this value if the CDR device is connected over a TCP/IP link, and this link is defined as either CDR1 or CDR2 on the IP Services screen.

Primary Output Extension

This field determines where the switch sends the CDR records, and is required if you specify a Primary Output Format.

Valid entries	Usage
eia	If you use the EIA port to connect the CDR device, enter **eia**. This is not a valid option on DEFINITY Server R systems.
Extension number	This is the extension of the data module (if used) that links the primary output device to the switch.

Secondary Output Format

Controls the format of the call records sent to the secondary output device.

⚠ **CAUTION:**

Only qualified (Avaya) service personnel should administer a secondary output device. This option may cause loss of data when the buffer contains large amounts of data.

Valid entries	Usage
lsu **unformatted** **int-direct** **int-process**	These are the only formats you can use for a secondary output device. The format must be compatible with your call accounting software. Verify this through your vendor or the accounting system documentation.

Secondary Output Extension

Appears when the secondary output format is administered.

Valid entries	Usage
eia	Use this if the secondary output device is connected to the eia port. This is not a valid option on DEFINITY Server R systems.
Extension number	This is the extension of the data module (if used) that links the secondary output device to the switch.

Use ISDN Layouts

ISDN Layouts provide more accurate information about the inter-exchange carrier and ISDN network services used for a call. This affects “lsu” and “printer” output formats, as well as any format with ISDN layouts, such as “teleseer.”

Valid entries	Usage
y/n	Enter **y** to enable the use of the ISDN version of the specified primary output format. You cannot use ISDN formats and Enhanced formats at the same time.

EIA Device Bit Rate

Applies to DEFINITY Server SI only. Displays if either Primary or Secondary Output Format is **eia**.

Valid entries	
300 **1200** **2400** **9600**	Enter the baud rate of the CDR device connected to the EIA port.

Use Enhanced Formats

Enhanced formats provide additional information about time in queue and ISDN call charges, where available. This affects the “expanded”, “teleseer”, “lsu”, “printer”, and “unformatted” output formats.

Valid entries	Usage
y/n	Enter **y** to enable the use of the Enhanced version of the specified primary output format. You cannot use Enhanced formats and ISDN formats at the same time.

Modify Circuit ID Display

This affects the “printer,” “teleseer,” and “59-character” output formats.

Valid entries	Usage
y	Enter **y** to display the circuit ID in its actual format (100’s, 10’s, units). For example, circuit ID 123 displays as 123. You might need to verify that your output device can accept this format.
n	Enter **n** to display the circuit ID in its default format (10’s, units, 100’s). For example, circuit ID 123 appears as 231.

Remove # From Called Number

Valid entries	Usage
y	Enter **y** to have the “#” (or “E”) symbol removed from the Dialed Number field of the call detail record. You might need to verify that your output device can accept this format.
n	Enter **n** to have the trailing “#” (or “E”) symbol appear in the Dialed Number field whenever inter-digit time out occurs or users dial # to indicate the end of dialing.

Record Outgoing Calls Only

Valid entries	Usage
y	Enter **y** to record only outgoing calls. This can save space if you are only concerned with charges for outbound calls.
n	Enter **n** to record both outgoing and incoming calls.

Intra-Switch CDR

Valid entries	Usage
y/n	Enter **y** to record calls within the switch. If you choose this option, you must complete the Intraswitch CDR screen to indicate which extensions to monitor.

Suppress CDR for Ineffective Call Attempts

Ineffective call attempts are calls that are blocked because the user did not have sufficient calling privileges or because all outgoing trunks were busy. This includes the unavailable incoming or outgoing trunks due to trunk usage allocation for ISDN Call-by-Call Service Selection trunks, incoming calls rejected by the switch due to NSF mismatch, and ISDN calls that did not complete at the far end, if a cause value was provided. These calls appear on the CDR record with a condition code “E.”

Valid entries	Usage
y	Enter **y** to ignore ineffective call attempts. Use this if you have limited storage space for CDR records and records often overrun the buffer.
n	Enter **n** to report ineffective call attempts. This can tell you if your users are often unable to place outgoing calls, or if a large number of incoming calls are not completed. You can also use this if you need to have records of attempts to contact a client, and are using ISDN trunks. Using this option requires more space for records.

Outg Trk Call Splitting

See ‘‘Call Splitting’’ on page 1439 for more information.

Valid entries	Usage
y/n	Enter **y** to create separate records for each portion of outgoing calls that are transferred or conferenced.

Disconnect Information in Place of FRL

See ‘‘Call detail record field descriptions’’ on page 1471 for more information.

Valid entries	Usage
y	Enter **y** to replace the Facility Restriction Level (FRL) field with information about why a call disconnects. You can use this information to isolate problems between the DEFINITY Server R and the telephone network.
n	Enter **n** to record the call’s FRL.

Outg Attd Call Record

Only appears if Outg Trk Call Splitting is **y**.

Valid entries	Usage
y/n	Enter **y** to enable separate recording of attendant portions of outgoing calls that are transferred or conferenced.

Interworking Feat-flag

See “Call detail record field descriptions’’ on page 1471 for more information.

Valid entries	Usage
y	Enter **y** if you want the feature flag to indicate interworked outgoing ISDN calls. An interworked call is one that passed through more than one ISDN node.
n	Enter **n** if you want the feature flag to indicate no answer supervision for interworked calls.

Force Entry of Acct Code for Calls Marked on Toll Analysis Screen

Specifies whether an account code will be required when making a toll call. This will not necessarily be all chargeable calls and it may even include some non-chargeable calls. See *“Forcing users to enter account codes’’ on page 530* for more information.

Valid entries	Usage
y	Enter **y** to deny all toll calls unless the user dials an account code. Forced Entry of Account Codes must be **y** on the System-Parameters Customer-Options screen.
n	Enter **y** to allow calls without an account code. This does not override other calling restrictions.

Calls to Hunt Group — Record

Valid entries	Usage
member-ext	Enter **member-ext** to record the extension of the phone or data terminal where the call terminated.
group-ext	Enter **group-ext** to record the extension that was dialed.

Record Called Vector Directory Number Instead of Group or Member

If this option is enabled, the called VDN overrides the group or member information that normally appears in the Dialed Number Field of the CDR record. If a call is directed through more than one VDN, the first VDN used for the call is stored. This applies only to calls routed to a hunt group by a vector, not to calls routed directly to an extension by a vector.

You cannot use both the Called VDN and the Agent Login ID Instead of Group or Member. Only one of these fields can be **y**.

Valid entries	Usage
y/n	Enter **y** to include the Vector Directory Number (VDN) in the Dialed Number Field of a CDR record.

Record Called Agent Login ID Instead of Group or Member

Only displays if Expert Agent Selection (EAS) is enabled on the System-Parameters Customer-Options screen. You cannot use both the Called VDN and the Agent Login ID Instead of Group or Member. Only one of these fields can be **y**.

Valid entries	Usage
y/n	Enter **y** to include the EAS agent’s LoginID instead of the physical extension in the Dialed Number Field of a CDR record.

Inc Trk Call Splitting

Appears when the Record Outgoing Calls Only field on the System Parameters CDR screen is **n**.

Valid entries	Usage
y/n	Enter **y** to create separate records for each portion of incoming calls that are transferred or conferenced.

Inc Attd Call Record

Appears when Inc Trk Call Splitting is **y**.

Valid entries	Usage
y/n	Enter **y** to enable separate recording of attendant portions of outgoing calls that are transferred or conferenced.

Record Non-Call-Assoc TSC

A temporary signaling channel (TSC) is a virtual connection established within an ISDN D-channel. For more information, see the *DS1/CEPT1/ISDN PRI Reference*.

Valid entries	Usage
y/n	Enter **y** to create records for non-call-associated temporary signaling connections. If you have a lot of data connections this could increase the number of records. You may want to consider the capacity of your record collection device.

Call Record Handling Option

Displays on DEFINITY Server R only. Used to control call routing when new calls come in, the CDR link is down, and the buffer is filled.

NOTE:
Changing this field from the default **warning** may cause ACD & vector calls that are measured by CDR to be redirected.

Valid entries	Usage
reorder	Enter **reorder** to block calls with reorder tone when the buffer is full. This applies to all calls. If you choose this option, no one will be able to make or receive calls if CDR is unable to record them.
warning	Enter **warning** to stop call recording when the buffer is full. This generates a minor alarm.
attendant	Enter **attendant** to route all calls to the attendant as non-CDR calls.

Record Call-Assoc TSC

Valid entries	Usage
y/**n**	Enter **y** to create records for call-associated temporary signaling connections. If you have a lot of data connections this could increase the number of records. You may want to consider the capacity of your call collection device.

Digits to Record for Outgoing Calls

Valid entries	Usage
dialed	Use **dialed** to record the digits a user actually dials.
outpulsed	Use **outpulsed** to record the digits that the switch actually sends out over the trunk, including any additions or deletions that take place during routing.

Privacy — Digits to Hide

If you enable CDR Privacy on the Station screen for a given phone, use this field to indicate how much of the dialed number to hide on the CDR record.

Valid entries	Usage
0–**7**	Enter the number of digits to hide, counting from the end (right to left). For example, if you enter **4** in this field and the user dials 555-1234, only “555” would appear in the Dialed Number field of the CDR record.

CDR Account Code Length

Valid entries	Usage
1–**15**	Enter the number of digits to record when a user enters an account code. For some record formats, a long account code overwrites spaces on the record that are usually assigned to other fields.

Field descriptions for page 2

```
change system-parameters cdr
                                  CDR SYSTEM PARAMETERS                    Page  2 of  2
       Data Item - Length               Data Item - Length                 Data Item - Length
   1: time____________ - 4_  17: ________________ - __  33: ________________ - __
   2: space___________ - 1_  18: ________________ - __  34: ________________ - __
   3: duration________ - 4_  19: ________________ - __  35: ________________ - __
   4: return__________ - 1_  20: ________________ - __  36: ________________ - __
   5: line-feed_______ - 1_  21: ________________ - __  37: ________________ - __
   6: ________________ - __  22: ________________ - __  38: ________________ - __
   7: ________________ - __  23: ________________ - __  39: ________________ - __
   8: ________________ - __  24: ________________ - __  40: ________________ - __
   9: ________________ - __  25: ________________ - __  41: ________________ - __
  10: ________________ - __  26: ________________ - __  42: ________________ - __
  11: ________________ - __  27: ________________ - __  43: ________________ - __
  12: ________________ - __  28: ________________ - __  44: ________________ - __
  13: ________________ - __  29: ________________ - __  45: ________________ - __
  14: ________________ - __  30: ________________ - __  46: ________________ - __
  15: ________________ - __  31: ________________ - __  47: ________________ - __
  16: ________________ - __  32: ________________ - __  48: ________________ - __
                                 Record length = 11
```

Screen 56. CDR System Parameters

This page appears only if Primary Record Format is customized.

Data Item

Enter the data items in the order they should appear on the customized record. Only use this screen if you have arranged with your vendor to customize your call accounting system to receive these records.

You must include at least one field in order to have a record. See the table below for valid entries. The last two data items in a the record must be **line-feed** and **return**, in that order.

For more information, see “Call detail record field descriptions” on page 1471.

Table 14. Valid Data Item entries

Data Item	Length	Data Item	Length
acct-code	15	ins	3
attd-console	2	isdn-cc	11
auth-code	7	ixc-code	4

Continued on next page

Table 14. Valid Data Item entries (Continued)

Data Item	Length	Data Item	Length
bandwidth	2	line-feed	1
bcc	1	ma-uui	1
calling-num	15	node-num	2
clg-pty-cat	2	null	1
clg-num-in-tac	10	null	1
code-dial	4	out-crt-id	3
code-used	4	ppm	5
cond-code	1	res-flag	1
date	6	return	1
dialed-num	23	sec-dur	5
duration	4	space	1
feat-flag	1	time	4
frl	1	**tsc_ct**	**4**
in-crt-id	3	tsc_flag	1
in-trk-code	4	vdn	5

Length

Enter the length of each data item, if different from the default.

Valid entries	Usage
The maximum record length depends on the call accounting system you use. Check with your vendor.	The date field should be six-digits to ensure proper output. Certain fields default to the required length.

Record Length

Displays the accumulated total length of the customized record, updated each time the length of a data item changes.

Class of Restriction

Use this screen to establish classes of restriction (COR). Classes of restriction control call origination and termination. Your system may use only one COR or as many as necessary to control calling privileges. You can assign up to 96 different CORs (0 – 95).

Consider the following to enhance your system security:

1. Assign a separate COR to incoming and outgoing trunk groups, then restrict calling between the two groups.
2. Limit the calling permissions as much as possible by setting appropriate Calling Party Restrictions and Facility Restriction Levels (FRLs).

Field descriptions for page 1

```
change cor 10                                                       Page 1 of 4
                       CLASS OF RESTRICTION

               COR Number: 10
          COR Description: supervisor

                      FRL: 0                                    APLT? y
   Can Be Service Observed? n                Calling Party Restriction: none
Can Be A Service Observer? y                  Called Party Restriction: none
        Time of Day Chart: 1             Forced Entry of Account Codes? n
         Priority Queuing? n                       Direct Agent Calling? y
     Restriction Override: none            Facility Access Trunk Test? n
     Restricted Call List? n                        Can Change Coverage? n
   Unrestricted Call List? __ __ __ __ __ __ __ __ __ __
            Access to MCT? y                  Fully Restricted Service? n
Group II Category For MFC: 7                  Hear VDN of Origin Annc.? n
         Send ANI for MFE? n_                   Add/Remove Agent Skills? y
            MF ANI Prefix: ______             Automatic Charge Display? n
Hear System Music on Hold? y      PASTE(Display PBX Data on telephone)? n
                              Can Be Picked Up By Directed Call Pickup? n
                                          Can Use Directed Call Pickup? n
                                          Group Controlled Restriction: inactive
```

Screen 57. Class of Restriction

COR Number

This is a display-only field when the screen is accessed via an administration command such as **change** or **display**.

COR Description

Valid entries	Usage
Up to 35 characters	Enter a description of the COR that indicates how you use it. If you make this as clear as possible (for example, Customer Service, Legal Department), it will be easier to remember which COR to assign when you add users.

FRL

Valid entries	Usage
0 to **7**	Enter an originating FRL number. AAR and/or ARS features use this entry to determine call access to an outgoing trunk group. Outgoing call routing is determined by a comparison of the FRLs in the AAR/ARS Routing Pattern and the FRL associated with the COR of the call originator (typically, a telephone user). An originating FRL of 0 has the least calling privileges. To enhance system security, assign the lowest possible FRL.

APLT

Valid entries	Usage
y/n	Enter **n** to allow access to APLT trunk group Enhanced Private Switched Communications System (EPSCS) or Common Control Switched Arrangement (CCSA) off-net facilities. If fully restricted service is enabled, set this field to **n**.

Can Be Service Observed

Note that this field allows or denies service observing for not only physical extensions, but also for logical agent IDs and VDNs. If you want an observer to observe users, set the users' CORs to **y** on the observer's COR Service Observing Permission table.

Valid entries	Usage
y/n	Enter **y** if users with this COR can be service observed.

Calling Party Restriction

This field determines the level of calling restriction associated with this COR.

NOTE:
To enhance system security, limit calling permissions as much as possible.

Valid entries	Usage
Origination	Blocks the calling party from originating a call from the facility at any time. The party can only receive calls. A phone with this COR may initiate Remote Access calls, if the COR of the barrier code allows it.
Outward	Blocks the calling party from calling outside the private network. Users can dial other users on the same switch or within a private network. To enhance security, Avaya recommends that you use outward restrictions when practical.
All-toll	Blocks the calling party from making ARS and trunk access calls from a facility assigned the COR to certain toll areas as defined in the Dialed String field on the Toll Analysis screen. The Dialed String field must be marked as being associated with the system's Toll List. The call completes if the facility's COR also is associated with an Unrestricted Call List and whose Dialed String field also matches the dialed number.
Tac-toll	Blocks the calling party from making trunk access calls from the facility assigned the COR to certain toll areas as defined in the Dialed String field on the Toll Analysis screen. The Dialed String field must be marked as being associated with the system's Toll List. The call completes if the facility's COR also is associated with an Unrestricted Call List and whose Dialed String field also matches the dialed number. See ''Toll Analysis'' on page 1163 for additional information.
None	No calling party restrictions.

Can Be a Service Observer

If you want an observer to observe users, set the users' CORs to **y** on the observer's COR Service Observing Permission table.

SECURITY ALERT:

The use of Service Observing features may be subject to federal, state, or local laws, rules, or regulations; or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves with and comply with all applicable laws, rules, and regulations before using these features.

NOTE:
You cannot enter **y** in the previous two fields unless Service Observing (Basic) is enabled on the System-Parameters Customer-Options screen.

Valid entries	Usage
y/n	Enter **y** if users with this COR can service observe other users.

Called Party Restriction

Valid entries	Usage
Inward	Blocks the calling party from receiving incoming exchange network calls, attendant originated calls, and attendant completed calls.
Manual	Blocks the called party from receiving all calls except for those originated or extended by the attendant.
Public	Blocks the called party from receiving public network calls. Attendant calls are allowed to go through to the called party as well as attendant-assisted calls if the Restriction Override field in the public restricted station's COR is **attd** or **all**.
Termination	Blocks the called party from receiving any calls at any time.
None	No called party restrictions.

Partitioned Group Number

This field appears only if AAR/ARS Partitioning is **y** and Time of Day Routing is **n** on the System Parameters Customer-Options screen.

Valid entries	Usage
1–**8**	Enter the AAR/ARS partitioned group number associated with this COR.

Time of Day Chart

Appears only if Time of Day field is enabled on the System Parameters Customer-Options screen. Refer to *"Setting up time of day routing'' on page 264* for more information.

Valid entries	Usage
1–**8**	Enter the AAR/ARS time-of-day-chart number associated with this COR.

Forced Entry of Account Codes

FEAC must be enabled on the System Parameters Customer-Options screen and on the CDR System Parameters screen.

Refer to ''Forced Entry of Account Codes'' on page 1438 and ''Forcing users to enter account codes'' on page 530 for more information.

NOTE:
If a COR requiring entry of account codes is assigned a VDN, the route to commands executed by the associated vector will not be successful.

Valid entries	Usage
y/n	Enter **y** to indicate that an account code must be dialed when making outgoing trunk calls. If this is **y**, any telephone assigned the associated COR must dial an account code before making an outgoing call. If you set this to y for a COR assigned to a trunk group, users must dial account codes before calling out over that trunk group. This may be useful for trunks used in international calls, and those that are more expensive. If a call is being routed by ARS, account code checking is not done on the COR.

Priority Queuing

Valid entries	Usage
y	Enter **y** to allow the telephone user's calls to be placed ahead of non-priority calls in a hunt group queue
n	If you do not use Automatic Call Distribution (ACD is not enabled on the System-Parameters Customer-Options screen), this field must be **n**.

Direct Agent Calling

Valid entries	Usage
y/n	If this is **y**, users may dial an ACD agent's extension directly, rather than anyone in the agent pool. If the system is in Night Service, the call routes to the Night Service extension. If the extension with this COR belongs to an agent, the agent may receive calls directly.

Restriction Override

Allows the specified users to bypass restriction on conference, transfer or call forwarding operations.

Valid entries	Usage
attendant	A telephone with a COR that is inward restricted cannot receive public network, attendant-originated, or attendant-extended calls. Enter **attendant** to give your attendants the ability to override this restriction.
all	Enter **all** if you want all of the users with this COR to override inward restrictions.
none	Enter **none** if you do not want any users of this COR to bypass the restrictions.

Facility Access Trunk Test

An associated feature button ("trk-ac-alm") status lamp lights when a successful test attempt occurs. Pressing one of the alarm buttons (ten maximum) when its associated status lamp is lit turns off all lamps on all buttons whether the access is still in progress or has completed.

Valid entries	Usage
y/n	Enter **y** to allow users with this COR to perform Facility Access Trunk Tests.

Restricted Call List

This list can be used whether the COR is toll restricted. The Restricted Call List (RCL) has priority over the Toll Analysis Unrestricted Call List (UCL). A call attempt from a facility assigned a COR (with RCL field set to **y**), whose dialed digit string is on the Toll Analysis screen and is marked as being associated with the RCL, will be denied.

Valid entries	Usage
y/n	Enter **y** to specify that this COR will have access to the system's Restricted Call List (see ''Toll Analysis'' on page 1163).

Can Change Coverage

Valid entries	Usage
y/n	Enter **y** to allow station users with this COR to select one of two previously administered coverage paths or to activate, change, or deactivate call forward all calls or call forward busy/don't answer from any on-site or off-site location.

Unrestricted Call List

Any entry on the Toll Analysis screen with an "X" in the Toll List column is restricted, meaning that the system blocks any attempt to complete a call containing the Dialed String. However, this field overrides that restriction.

For example, if the Toll Analysis screen shows a Dialed String entry of 538 and there is an "X" in the Toll List column, the 538 number is restricted. To override this restriction, in the Toll Analysis screen, enter **X** in the "5" column under the Unrestricted Call List heading. In the Class of Restriction screen, in this field, enter **5** to complete the restriction override.

Valid entries	Usage
1–10	Appears when Calling Party Restriction is **all-toll** or **tac-toll**. This field allows a user to complete a toll call with "restricted" dialed digits. This field is associated with the Dialed String field on the Toll Analysis screen. An Unrestricted Call List number is denoted on that screen.

Access to MCT?

This field refers to Malicious Call Trace.

Valid entries	Usage
y	Enter **y** to allow permissions to activate a request to trace a malicious call.
n	Entering **n** prohibits this user from requesting a malicious call trace, but does not prevent this extension from appearing in the MCT History report, should this extension be the subject of a malicious call trace.

Fully Restricted Service

NOTE:
If this field is enabled, the APLT field must be **n**.

Valid entries	Usage
y/n	When **y** entered for a given COR, stations assigned that COR will not have access to the public network for either incoming or outgoing calls.

Use COR for all Group II Responses

This field only appears if the Outgoing Call Type field is set to group-ii-mfc.

Valid entries	Usage
y/n	Y allows the COR administered category to be used for both the calling party and called party categories.

Group II Category For MFC

This field always controls categories for Russian signaling trunks. It can control categories for R2-MFC signaling trunks, depending on the value of the Use COR for Calling Party Category field on the Multifrequency-Signaling-Related System Parameters screen.

The Calling Party Category digit administered in this field is included as part of the ANI information sent to the Central Office on request using R2-MFC signaling.

Valid entries	Usage
1 –**10**	Enter the value you want the switch to send as the Calling and/or Called Party Category for phones or trunks that use this COR.

Category For MFC ANI

The Category for MFC ANI field always controls categories for Russian signaling trunks. It also may control categories for R2-MFC signaling trunks, depending on what value is in the Use COR for Calling Party Category field on the system-parameters multi-frequency screen.

The Calling Party Category digit administered in this field is included as part of the ANI information sent to the Central Office on request using R2-MFC signaling.

Valid entries	Usage
1 –**10**	Used in other than U.S.

Hear VDN of Origin Announcement

Valid entries	Usage
y/n	Enter **y** if users with this COR can receive VDN of Origin messages.

Send ANI for MFE

Only applicable for Spain. Valid for 2/6 signaling, but not 2/5 signaling. The following field appears only if Expert Agent Selection (EAS) is enabled on the Feature-Related System-Parameters screen.

Valid entries	Usage
y	Enter **y** to enable Automatic Number Identification (ANI). When the value is **y**, the switch sends the calling party’s number to the public or IBERCOM network so that charges will be broken down by line.
n	If this value is **n**, charges are not itemized by line, and your company will receive a single bill for the total number of calls made (block charging).

Add/Remove Agent Skills

Valid entries	Usage
y/n	Enter **y** to allow users with this COR to add and remove skills.

MF ANI Prefix

Defines the prefix to apply to an extension number when ANI is sent to the CO. This overrides any ANI prefix administered on the Multifrequency Signaling screen. This does not apply when ANI is tandemed through the switch on tandem calls. This field also applies to the ANI for the switch when the originating side is a trunk and there was no ANI.

Valid entries	Usage
1–7 digits	If you want the entire number to display on the receiving end, enter all digits except the extension number.

Hear System Music on Hold

Valid entries	Usage
y/n	Enter **y** to allow the Music on Hold feature to be activated by a telephone.

PASTE (Display PBX Data on telephone)

Valid entries	Usage
y/n	Enter **y** to download all lists. Enter **n** to disallow the PASTE feature.

Automatic Charge Display

Shows the cost of an active outgoing call using Periodic Pulse Metering (PPM) or ISDN Advice of Charge (AOC) on Digital Communications Protocol (DCP) or Avaya BRI stations. Not available in the U.S.

Valid entries	Usage
y	Displays call charges during and at the end of the call.
n	Call charges can be seen if users press the DISP-CHRG button before the call drops.

Can Be Picked Up By Directed Call Pickup

Valid entries	Usage
y/n	Enter **y** to allow this Station's or EAS agent's calls to be picked up by using the Directed Call Pickup Up feature. Before you can set this field to y, you must set Directed Call Pickup on the Feature-Related System Parameters screen to **y**.

Can Use Directed Call Pickup

Valid entries	Usage
y/n	Enter **y** to allow the station, attendant, or EAS agent to pick up calls using the Directed Call Pickup feature. Set Directed Call Pickup on the Feature-Related System Parameters screen to **y** to set this field to **y**.

Group Controlled Restriction

Determines if the current COR is under controlled restriction. This field can help troubleshoot problems by first checking its value.

Valid entries	Usage
active	indicates the COR is controlled restricted
inactive	indicates the COR is not controlled restricted

Field descriptions for page 2

```
change cor 1                                                     Page 2 of 4
                         CLASS OF RESTRICTION

                     MF Incoming Call Trace? n
               Brazil Collect Call Blocking? n
                     Block Transfer Display? n
 Block Enhanced Conference/Transfer Displays? y
                           Station Lock COR: 10
```

Screen 58. Class of Restriction screen

MF Incoming Call Trace

Valid entries	Usage
y/n	Enter **y** to allow assignment of a Call Trace COR to a station. Avaya MultiVantage then generates an MFC backward signal (administered on the System-Parameters Multifrequency-Signaling screen) during call setup instead of the "free" signal. This triggers the central office to collect trace information before releasing the calling party, if the terminating station's COR has this feature set to **y**.

Brazil Collect Call Blocking

For Brazil only

Valid entries	Usage
y/n	Enter **y** to permit all Brazilian trunks calls that terminate to a station to send back a double answer to the CO. This double answer tells the CO that this particular station cannot accept collect calls. The CO then tears down the call if it is a collect call. Set Country on the trunk group screen to **23** and set this field to **y.**

Block Transfer Display

Valid entries	Usage
y/n	Enter **y** to prevent users of DCP, Hybrid, ISDN-BRI, or wireless display telephones from receiving a confirmation message when they transfer a call.

Block Enhanced Conference/Transfer Display

Use this field to add display messages regarding conference and transfer features on digital phones.

Valid entries	Usage
y/n	Enter **y** to block all the enhanced conference/transfer display messages except "Transfer Completed"

Station Lock COR

This field defaults to the current screen COR. Extensions that are assigned this COR can use Station Lock with the access code administered on the FAC screen.

Valid entries	Usage
0 - 95	This field defaults to current COR.

Remote Logout of Agent

Use a feature access code to logout an idle ACD or EAS agent without being at the agent's phone.

Valid entries	Usage
y/n	Enter **y** to allow remote logout of an idle ACD or EAS agent.

Field descriptions for page 3

```
change cor 10                                                Page 3 of 4
                      CLASS OF RESTRICTION
CALLING PERMISSION (Enter y to grant permission to call specified COR)
  0? n     12? n     24? n     36? n     48? n     60? n    72? n     84? n
  1? n     13? n     25? n     37? n     49? n     61? n    73? n     85? n
  2? n     14? n     26? n     38? n     50? n     62? n    74? n     86? n
  3? n     15? n     27? n     39? n     51? n     63? n    75? n     87? n
  4? n     16? n     28? n     40? n     52? n     64? n    76? n     88? n
  5? n     17? n     29? n     41? n     53? n     65? n    77? n     89? n
  6? n     18? n     30? n     42? n     54? n     66? n    78? n     90? n
  7? n     19? n     31? n     43? n     55? n     67? n    79? n     91? n
  8? n     20? n     32? n     44? n     56? n     68? n    80? n     92? n
  9? n     21? n     33? n     45? n     57? n     69? n    81? n     93? n
 10? n     22? n     34? n     46? n     58? n     70? n    82? n     94? n
 11? n     23? n     35? n     47? n     59? n     71? n    83? n     95? n
```

Screen 59. Class of Restriction screen

CALLING PERMISSION

Valid entries	Usage
y/n	A **y** means an originating facility assigned this COR can be used to call facilities assigned this COR. Enter **n** for each COR number (0 through 95) that cannot be called by the COR being implemented.

Field descriptions for page 4

```
change cor 10                                                  Page 4 of 4
                      CLASS OF RESTRICTION
SERVICE OBSERVING PERMISSIONS
(Enter y to grant permission to service observe specified COR)
  0? n     12? n     24? n     36? n     48? n     60? n    72? n     84? n
  1? n     13? n     25? n     37? n     49? n     61? n    73? n     85? n
  2? n     14? n     26? n     38? n     50? n     62? n    74? n     86? n
  3? n     15? n     27? n     39? n     51? n     63? n    75? n     87? n
  4? n     16? n     28? n     40? n     52? n     64? n    76? n     88? n
  5? n     17? n     29? n     41? n     53? n     65? n    77? n     89? n
  6? n     18? n     30? n     42? n     54? n     66? n    78? n     90? n
  7? n     19? n     31? n     43? n     55? n     67? n    79? n     91? n
  8? n     20? n     32? n     44? n     56? n     68? n    80? n     92? n
  9? n     21? n     33? n     45? n     57? n     69? n    81? n     93? n
 10? n     22? n     34? n     46? n     58? n     70? n    82? n     94? n
 11? n     23? n     35? n     47? n     59? n     71? n    83? n     95? n
```

Screen 60. Class of Restriction

SERVICE OBSERVING PERMISSION

Valid entries	Usage
y/n	A **y** grants permission to observe specific CORs. Enter **n** for each COR number (0 through 95) that cannot be observed by the COR being implemented.

Class of Service

This screen administers access permissions for call processing features that require dial code or feature button access.

NOTE:
Class of Service (COS) does not apply to trunk groups except for the Remote Access feature.

A COS assignment defines whether or not a telephone user may access or use the following features and functions. Up to 16 different COS numbers may be administered (0–15).

```
change cos                                                            Page 1 of 1
                                   CLASS OF SERVICE

                                        0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15

    Auto Callback                       n y y n y n y n y n  y  n  y  n  y  n
    Call Fwd-All Calls                  n y n y y n n y y n  n  y  y  n  n  y
    Data Privacy                        n y n n n y y y y n  n  n  n  y  y  y
    Priority Calling                    n y n n n n n n n y  y  y  y  y  y  y
    Console Permissions                 n n n n n n n n n n  n  n  n  n  n  n
    Off-hook Alert                      n n n n n n n n n n  n  n  n  n  n  n
    Client Room                         n n n n n n n n n n  n  n  n  n  n  n
    Restrict Call Fwd-Off Net           n y y y y y y y y y  y  y  y  y  y  y
    Call Forward Busy/DA                n n n n n n n n n n  n  n  n  n  n  n
    Personal Station Access             n n n n n n n n n n  n  n  n  n  n  n
    Extended Forwarding All             n n n n n n n n n n  n  n  n  n  n  n
    Extended Forwarding B/DA            n n n n n n n n n n  n  n  n  n  n  n
    Trk-to-Trk Restriction Override     n n n n n n n n n n  n  n  n  n  n  n
    QSIG Call Offer Originations        n n n n n n n n n n  n  n  n  n  n  n
    Automatic Exclusion                 n n n n n n n n n n  n  n  n  n  n  n
```

Screen 61. Class of Service screen

The screen lists the default values for each COS/feature combination. For a particular combination, **y** allows access to the feature and **n** denies access. Assign entries on the screen for each COS to be implemented. Default values are shown on the screen.

CAUTION:
Because many hunt groups are set up with COS 1, be careful when you assign restrictions to COS 1.

Automatic Callback

Allows this user to request Automatic Callback. Refer to ''Automatic Callback'' on page 1370 for more information.

Call Forwarding All Calls

Allows this user to forward all calls to any extension. See ''Call Forwarding'' on page 1496 for more information.

Data Privacy

Allows this user to enter a feature access code to protect a data call from interruption by any of the system's override or ringing features. See ''Data Privacy'' on page 490 for more information.

Priority Calling

Allows user to dial a feature access code to originate a priority call. Such calls ring differently and override send all calls, if active. See ''Priority Calling'' on page 1677 for more information.

Console Permissions

Console Permissions allow multiappearance telephone users to control the same features that the attendant controls. You might assign this permission to front-desk personnel in a hotel or motel, or to a call center supervisor. With console permission, a user can:

- Activate Automatic Wakeup for another extension
- Activate and deactivate controlled restrictions for another extension or group of extensions
- Activate and deactivate Do Not Disturb for another extension or group of extensions
- Activate Call Forwarding for another extension
- Add and remove agent skills
- Record integrated announcements

Off-Hook Alert

See ''Emergency Access to the Attendant'' on page 1548 for more information. To enable this option, either the Hospitality (Basic) or Emergency Access to Attendant field must be enabled on the System-Parameters Customer-Options screen.

Client Room

Allows users to access Check-In, Check-Out, Room Change/Swap, and Maid status functions. In addition, Client Room is required at consoles or telephones that are to receive message-waiting notification. You can administer class of service for Client Room only when you have Hospitality Services and a Property Management System interface. See *GuestWorks and DEFINITY Systems Technician Handbook for Hospitality Installations* for more information.

Restrict Call Fwd-Off Net

This restricts users from forwarding calls to the public network. For security reasons, this should be enabled for all classes of service except the ones you use for very special circumstances. See ‘‘Call Forwarding Off Net’’ on page 1497 for more information.

Call Forwarding Busy/DA

Allows this user to forward calls to any extension when the dialed extension is busy or does not answer. See ‘‘Call Forwarding’’ on page 1496 for more information.

Personal Station Access

Allows users to associate a telephone to their extension with their programmed services, using a feature access code. This field must be set to n for virtual phones. You cannot change this field to **y** if Personal Station Access (PSA) on the System Parameters Customer-Options screen is **n**. See ‘‘Personal Station Access’’ on page 1674 for more information.

Extended Forwarding All

Allows a user to administer call forwarding (for all calls) from a remote location. You cannot change a COS to **y** if Extended Cvg/Fwd Admin on the System Parameters Customer-Options screen is **n**. See ‘‘Extended User Administration of Redirected Calls’’ on page 1429 for more information.

Extended Forwarding B/DA

Allows this user to administer call forwarding (when the dialed extension is busy or does not answer) from a remote location. You cannot change this COS to **y** if Extended Cvg/Fwd Admin on the System Parameters Customer-Options screen is **n**. See ‘‘Extended User Administration of Redirected Calls’’ on page 1429 for more information.

Trk-to-Trk Restriction Override

Users with this COS override any system and/or COR-to-COR calling party restrictions that would otherwise prohibit the trunk-to-trunk transfer operation for users with this COS. See ''Transfer — Trunk-to-Trunk'' on page 1778 for more information.

SECURITY ALERT:

Use this COS capability with caution. The ability to perform trunk-to-trunk transfers greatly increases the risk of toll fraud.

QSIG Call Offer Originations

Allows this user to invoke QSIG Call Offer services. See *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

Automatic Exclusion

Allows a user to activate automatically Exclusion when they go off hook on a station that has an assigned EXCLUSION button. If set to **n**, allows a user manual exclusion when they press the EXCLUSION button before dialing or during a call. Appears when, on the Feature-Related System Parameters screen, the Automatic Exclusion by COS field is **y.**

Code Calling IDs

On systems with chime paging, use this screen to assign a unique series of chimes (a *chime code*) to extensions. The chime code assigned to an extension plays over the speakers whenever that extension is paged. You may assign chime codes to up to 125 extensions.

```
change paging code-calling-ids                                     Page 1 of 2

                                      CODE CALLING IDs
     ID ASSIGNMENTS
              Id   Ext          Id   Ext         Id   Ext         Id   Ext         Id   Ext
             111: ____         141: ____        221: ____        251: ____        331: ____
             112: ____         142: ____        222: ____        252: ____        332: ____
             113: ____         143: ____        223: ____        253: ____        333: ____
             114: ____         144: ____        224: ____        254: ____        334: ____
             115: ____         145: ____        225: ____        255: ____        335: ____
             121: ____         151: ____        231: ____        311: ____        341: ____
             122: ____         152: ____        232: ____        312: ____        342: ____
             123: ____         153: ____        233: ____        313: ____        343: ____
             124: ____         154: ____        234: ____        314: ____        344: ____
             125: ____         155: ____        235: ____        315: ____        345: ____
             131: ____         211: ____        241: ____        321: ____        351: ____
             132: ____         212: ____        242: ____        322: ____        352: ____
             133: ____         213: ____        243: ____        323: ____        353: ____
             134: ____         214: ____        244: ____        324: ____        354: ____
             135: ____         215: ____        245: ____        325: ____        355: ____
```

Screen 62. Code Calling IDs

Ext

This field assigns extensions to chime codes. Only one extension can be assigned to each chime code.

Valid entries	Usage
An extension number	Enter a physical extension, not a VDN, to assign that extension to a code. Otherwise, leave this field blank.

Related topics

Refer to ‘‘Setting up chime paging over loudspeakers’’ on page 466 for instructions.

Refer to ‘‘Loudspeaker paging’’ on page 1633 for a description of the feature.

Command Permission Categories

Use this screen to administer a user's permissions associated with their login. When set to y, the permissions on this screen apply for the object that is not restricted. Use the second and third pages of the Command Permission Categories screen to restrict a user from any access to specified objects. To see pages 2 and 3, type **y** in the Additional Restrictions field.

Field descriptions for page 1

```
change permissions angi3                                          Page 1 of 3
                           COMMAND PERMISSION CATEGORIES
                             Login Name: angi3
   COMMON COMMANDS
                  Display Admin. and Maint. Data? n
                             System Measurements? n
                  System Mgmt Data Transfer Only? n
   ADMINISTRATION COMMANDS
         Administer Stations? n              Administer Features? n
           Administer Trunks? n           Administer Permissions? n
     Additional Restrictions? n

   MAINTENANCE COMMANDS
           Maintain Stations? n        Maintain Switch Circuit Packs? n
             Maintain Trunks? n       Maintain Process Circuit Packs? n
            Maintain Systems? n                Maintain Enhanced DS1? n
```

Screen 63. Command Permission Categories

Login Name

This display-only field shows the login to which these permissions apply.

Display Admin. and Maint. Data

Valid entries	Usage
y/n	Enter **y** to allow a user to use the display, list, monitor, status, and schedule commands and also change their own passwords and schedule reports.

System Measurements

This field only appears for DEFINITY Server SI systems.

Valid entries	Usage
y/n	Enter **y** to allow a user to use the list measurements commands.

System Mgmt Data Transfer Only

Valid entries	Usage
y/n	Enter **y** to indicate the switch is accessible by Network Administration.

Administer Stations

Valid entries	Usage
y/n	Enter **y** to allow a user to add, change, duplicate, or remove stations, data modules and associated features, such as abbreviated dialing, vectors, and routing tables.

Administer Features

Use caution when assigning this permission to a user.

Valid entries	Usage
y/n	Enter **y** to allow a user to administer feature-related parameters, such as coverage paths, class of service, class of restriction, system parameters, authorization codes, and security.

Administer Trunks

Give this permission only to users who are very familiar with these features.

Valid entries	Usage
y/n	Enter **y** to allow a user to administer AAR/ARS, trunk groups, remote access, and route patterns.

Administer Permissions

This permission only applies to super-user logins.

Valid entries	Usage
y/n	Enter **y** to allow a user to administer logins and command permissions.

Additional Restrictions

Use page 2 and 3 to add objects (up to 40) that this user cannot manipulate. If an object appears on the Additional Restrictions page, users cannot display, add, change, or do anything else with that object.

Valid entries	Usage
y/n	Enter **y** to create additional restrictions, and to have the second and third pages of this screen appear.

Maintain Stations

You can only enter a value in this field if the Station and Trunk MSP field is set to **y** on the System-Parameters Customer-Options screen**.**

Valid entries	Usage
y/n	Enter **y** to allow a user to perform station maintenance.

Maintain Switch Circuit Packs

You can only enter a value in this field if the Station and Trunk MSP field is set to **y** on the System-Parameters Customer-Options screen**.**

Valid entries	Usage
y/n	Enter **y** to allow a user to perform circuit pack maintenance.

Maintain Trunks

You can only enter a value in this field if the Station and Trunk MSP field is set to **y** on the System-Parameters Customer-Options screen**.**

Valid entries	Usage
y/n	Enter **y** to allow a user to perform trunk maintenance.

Maintain Process Circuit Packs

You can only enter a value in this field if the Processor and System MSP field is set to **y** on the System-Parameters Customer-Options screen**.**

Valid entries	Usage
y/n	Enter **y** to allow a user to perform processor maintenance.

Maintain Systems

You can only enter a value in this field if the Processor and System MSP field is set to **y** on the System-Parameters Customer-Options screen**.**

Valid entries	Usage
y/n	Enter **y** to allow a user to perform system maintenance.

Maintain Enhanced DS1

You can only enter a value in this field if the DS1 MSP field is set to **y** on the System-Parameters Customer-Options screen**.**

Valid entries	Usage
y/n	Enter **y** to allow a user to perform enhanced DS1 maintenance.

Field descriptions for page 2

```
change permissions angi3                                        Page 2 of 3

                         COMMAND PERMISSION CATEGORIES
                            RESTRICTED OBJECT LIST
          ____________________        ___________________
          ____________________        ___________________
          ____________________        ___________________
          ____________________        ___________________
          ____________________        ___________________
          ____________________        ___________________
          ____________________        ___________________
          ____________________        ___________________
          ____________________        ___________________
          ____________________        ___________________
```

Screen 64. Command Permission Categories—Restricted Object List screen

Pages 2 and 3 of this screen are identical, and allow you to specify up to 40 objects that this user cannot access. These pages do not appear unless you set both the Administer Station and Additional Restrictions fields to **y** on page 1 of this screen. If you want to limit a user's permissions beyond those on page 1, enter the objects in this list. For example, if you want a user to be able to add and change stations, but not Vector Directory Numbers (VDN), in the Administer Stations and Additional Restrictions fields, type **y**. Then on page 2, type **vdn** as a restricted object.

Configuration Set

This screen defines a number of call treatment options for EC500 cell phone calls. The EC500 allows the use of up to 10 Configuration Sets, which are already defined in the system using default values.

Field descriptions for page 1

```
change xmobile configuration-set 1                              Page   1 of 1

                               CONFIGURATION SET: 1

  Configuration Set Description:
           Calling Number Style:
            CDR for Termination:
    Fast Connect on Origination?
   Post Connect Dialing Options?
```

Screen 65. Console Parameters — Default Attendant Group

Configuration Set Description

Describes the purpose of the configuration set.

Valid entries	Usage
Up to 20 characters	For example, EC500 handsets.

Calling Number Style

Determines the format of the caller ID for calls from a local switch extension to an EC500 cell phone.

Valid entries	Usage
network	Provides a display of only 10-digit numbers.
pbx	Provides a display of less than 10-digits.

CDR for Termination

Determines whether a call detail record is generated for any call to the cell phone.

Valid entries	Usage
station	The XMOBILE extension must be entered on the Intra-Switch CDR screen and be a standalone XMOBILE in order to get CDR reports.
trunk	Any call to the cell phone generates a CDR report.

Fast Connect on Origination

Determines whether some additional processing occurs on the switch prior to connecting a call..

Valid entries	Usage
y/n	Enter **y** to access this feature.

Post Connect Dialing Options

Determines whether additional capabilities, beyond standard ISDN dialing, are available for those incoming ISDN trunk calls that are maped into XMOBILE stations.

Valid entries	Usage
dtmf	For dual tone multiple frequencies.

Console Parameters

This screen administers attendant console group parameters. This includes basic parameters for Centralized Attendant Service (CAS) and Inter-PBX Attendant Service (IAS). A list of the administered attendant consoles also displays on this screen.

Field descriptions for page 1

```
change console-parameters                                         Page   1 of   4
                              CONSOLE PARAMETERS
         Attendant Group Name: OPERATOR
                          COS: 0                                   COR: 0
       Calls in Queue Warning: 5                   Attendant Lockout? y
        Ext Alert Port (TAAS):
                          CAS: none
                                             Night Service Act. Ext.:
                 IAS (Branch)? n             IAS Tie Trunk Group No.:
         IAS Att. Access Code:                 Alternate FRL Station:
              Backup Alerting? n       DID-LDN Only to LDN Night Ext? n
      Attendant Vectoring VDN:
```

Screen 66. Console Parameters — Default Attendant Group

Attendant Group Name

Enter a name for the attendant group.

COS

Enter a class of service (COS) number that reflects the desired features for all your attendant consoles. You can override this COS, by assigning a different COS on the individual Attendant screen.

COR

Enter the class of restriction (COR) number that reflects the desired features for the attendant. You can override this COR, by assigning a different COR on the individual Attendant screen.

Calls In Queue Warning

Enter the number of incoming calls that can be in the attendant queue before the console's second Call Waiting lamp lights. The console's first Call Waiting lamp lights when any incoming calls are waiting to be answered. The second lamp lights when the number of calls waiting equals the value you entered in the Calls in Queue Warning field.

Attendant Lockout

Attendant Lockout prevents an attendant from re-entering a multiple-party connection held on the console unless recalled by a telephone user.

Attendant Lockout provides privacy for parties on a multiple-party call held on the console. The held parties can hold a private conversation without interruption by the attendant.

Valid entries	Usage
y/n	Enter **y** to activate Privacy — Attendant Lockout. If **y** is entered, the attendant is prohibited from reentering a conference call that has been placed on hold unless recalled by a phone user on the call.

Ext Alert Port (TAAS)

Enter the seven-digit port number assigned to the external alerting device. This supports the Night Service — Trunk Answer From Any Station feature.

NOTE:
Type an "X" in this field to indicate that there is no hardware associated with this port assignment. If an **X** is used here, you must also fill in the Ext Alert (TAAS) Extension field.

Ext Alert (TAAS) Extension

Appears only when an **X** is entered in the Ext Alert Port (TAAS) field. This extension is used by the Terminal Translation Feature (TTI) to assign a port to the Ext Alert Port from a station on the Ext Alert port during system installation or provisioning. Once a port is assigned (either via TTI or by changing the Ext Alert Port field from the G3-MA or other manager terminal) the extension is automatically removed and treated as unassigned.

CAS

The CAS Main or Branch features must be enabled on the System- Parameters Customer-Options screen for either of these features to be functional here.

Valid entries	Usage
main	
branch	
none	
QSIG-main	Can be used if, on the System-Parameters Customer-Options screen, the Centralized Attendant field is **y**. Indicates all attendants are located on the main PBX.
QSIG-branch	Can be used if, on the System-Parameters Customer-Options screen, the Centralized Attendant field is **y**. Indicates there are no local attendants and routes to the main PBX.

RLT Trunk Group No.

Appears only when **branch** is entered in the CAS field. Enter the trunk group number corresponding to the Release Link Trunk (RLT) trunk group to the main location when supporting CAS Branch service.

CAS Back-Up Ext.

This field handles attendant-seeking calls if the RLT trunk group to the CAS Main switch is out of service or if CAS Back-Up is activated. This field must be explicitly defined as an extension in the dial plan. Neither a prefixed extension nor a VDN extension is allowed. Appears only when **branch** is entered in the CAS field.

Valid entries	Usage
An extension number for a station	
individual attendant console	
hunt group	
TEG	

AAR/ARS Access Code

Appears if the CAS field is **QSIG-branch**. An optional field that contains an AAR/ARS access code to route to the main PBX, if needed.

Valid entries	Usage
0 - 9, *****, **#**	Enter up to 4 digits.
blank	

Night Service Act. Ext.

This is a display-only field. It contains the extension of the current night service activation station, if any. Such a station is administered by assigning it a “night-serv” button.

IAS (Branch)

Enables or disables Inter-PBX Attendant Service (IAS) Branch feature. Does not appear if, on the System-Parameters Customer-Options screen, the Centralized Attendant field is **y**.

NOTE:
CAS and IAS cannot both be active at the same time.

QSIG CAS Number

Appears if the CAS field is **QSIG-branch**. Contains the complete number of the attendant group at the main switch, or a vector directory number (VDN) local to the branch switch. This field cannot be left blank.

Valid entries	Usage
0 - 9	Enter up to 20 digits.

IAS Tie Trunk Group No.

Enter the number of the tie trunk group to the main for the IAS (Branch). This entry is required when IAS Branch is **y**. Does not appear if, on the System-Parameters Customer-Options screen, the Centralized Attendant field is **y**.

IAS Att. Access Code

Enter the extension number of the attendant group at the main switch. This entry is required when IAS Branch is **y**. Does not appear if, on the System-Parameters Customer-Options screen, the Centralized Attendant field is **y**.

Alternate FRL Station

This is a display-only field. It displays the extension of the alternate facility restriction level (FRL) activation station.

Backup Alerting

Indicates whether or not system users can pick up alerting calls if the attendant queue has reached its warning state.

DID-LDN Only to LDN Night Ext.

Valid entries	Usage
y	Enter **y** to allow only listed directory number (LDN) calls to go to the listed directory night service extension.
n	Enter **n** if you want all attendant seeking calls to route to the LDN night service extension.

Attendant Vectoring VDN

This field appears only if, on the System-Parameters Customer-Options screen, the Attendant Vectoring field is **y** and the Tenant Partitioning field is **n**.

Valid entries	Usage
Assigned VDN extension or blank	Enter an assigned Attendant VDN extension or blank.

Field descriptions for page 2

```
change console-parameters                                        Page   2 of   4
                              CONSOLE PARAMETERS

TIMING
  Time Reminder on Hold (sec): 10           Return Call Timeout (sec): 10
  Time in Queue Warning (sec):

  INCOMING CALL REMINDERS
      No Answer Timeout (sec): 20                      Alerting (sec): 40
                               Secondary Alert on Held Reminder Calls? y

ABBREVIATED DIALING
     List1: group    1        List2:                   List3:
            SAC Notification? n
                           COMMON SHARED EXTENSIONS
            Starting Extension:                 Count:
```

Screen 67. Console Parameters — Default Attendant Group

Timed Reminder on Hold (sec)

Enter the time in seconds that a call remains on hold at the console before the attendant is alerted. In a CAS arrangement, the main and the branch consoles (when administered) should be administered the same.

Return Call Timeout (sec)

Enter the time in seconds before a split away call (call extended and ringing a station or otherwise split away from the console) returns to the console. Be sure to allow five seconds for each ring at all points in a coverage path to ensure the entire path is completed before the call returns to the console.

Time In Queue Warning (sec)

Enter the number of seconds a call can remain in the attendant queue before activating an alert.

No Answer Timeout (sec)

Enter the number of seconds a call to the attendant can remain unanswered without invoking a more insistent sounding tone. Be sure to allow five seconds for each ring at all points in a coverage path to ensure the entire path is completed before the call returns to the console.

Alerting (sec)

Enter the number of seconds after which a held or unanswered call is disconnected from an attendant loop and routed to another attendant or night service

Secondary Alert on Held Reminder Calls?

Valid entries	Usage
y	Enter **y** to begin attendant alerting for Held Reminder Calls with secondary alerting.
n	Enter **n** to have held reminder calls alert the attendant the same as normal calls. Normal calls start with primary alerting and switch to secondary alerting when the No Answer Timeout expires.

List1, List2, List3

You can assign up to 3 abbreviated dialing lists to each attendant. However, you cannot assign a personal list to an attendant.

Valid entries	Usage
enhanced	Allows the attendant to access the enhanced system abbreviated dialing list.
group	Allows the attendant to access the specified group abbreviated dialing list. You also must enter a group number.
system	Allows the attendant to access the system abbreviated dialing list.

SAC Notification

Enables or disables Enhanced Attendant Notification for Send All Calls.

Common Shared Extension—Starting Extension

These extension numbers can be used by the attendant to park calls.

Common Shared Extension—Count

Enter a number to indicate the number of consecutive extensions, beginning with the Start Extension to be used as common, shared extensions. For example, if you enter a starting extension of 4300 and a count of 3, the system provides three consecutive extension numbers (4300, 4301, and 4302) for parking calls.

The extensions should be assigned to the optional Attendant Selector Console in the 00 through 09 block (bottom row) in any hundreds group for easy identification by the attendant. The lamp associated with the number will identify “call parked” or “no call parked”, instead of busy or idle status.

Field descriptions for page 3

```
change console-parameters                                          Page 3 of 4
                                CONSOLE PARAMETERS
QUEUE PRIORITIES
            Emergency Access:1_
             Assistance Call:2_
                     CO Call:2_
            DID to Attendant:2_
                    Tie Call:2_
         Redirected DID Call:2_
             Redirected Call:2_
                 Return Call:2_
                 Serial Call:2_
 Individual Attendant Access:2_
             Interpositional:2_
    VIP Wakeup Reminder Call:2_
          Miscellaneous Call:2_

 Call-Type Ordering Within Priority Levels? n
```

Screen 68. Console Parameters — Default Attendant Group

Queue Priorities

Attendant Priority Queue allows attendants to answer calls by call category (for example, by trunk type). The Attendant Priority Queue handles incoming calls to an attendant when the call cannot be immediately terminated to an attendant. The calling party hears ringback until an attendant answers the call.

You may assign the same priority level to more than one call. Priority 1 is the highest priority and is the default for Emergency Access. Assign a priority level from **1** through **13** to each of the call types.

The attendant call categories are:

- Emergency Access — A call from a telephone user who dials the emergency access code (default is highest-priority level)
- Assistance Call— A call from a telephone user who dials the attendant-group access code, or from a telephone that has the Manual Originating Line Service feature activated
- CO Call — An incoming trunk call (CO/FX/WATS trunk) to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- DID to Attendant — An incoming DID trunk call to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.

- Tie Call — An incoming TIE trunk call (dial-repeating or direct types) to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- Redirected DID Call — A DID or ACD call that times out due to ring/no-answer, busy condition (if applicable), or Number Unobtainable and reroutes to the attendant group.
- Redirected Call — A call assigned to one attendant, but redirected to the attendant group because the attendant is now busy
- Return Call — A call returned to the attendant after it times out. If the attendant is now busy, the call redirects to the attendant group.
- Serial Call — A call from the Attendant Serial Call feature when an outside trunk call (designated as a serial call by an attendant) is extended to and completed at a telephone, and then the telephone user goes on-hook. If the attendant who extended the call is busy, the call redirects to the attendant group.
- Individual Attendant Access — A call from a telephone user, incoming trunk call, or a system feature to the Individual Attendant Access (IAA) extension of a specific attendant. If the attendant is busy, the call queues until the attendant is available.
- Interposition — A call from one attendant to the Individual Attendant Access (IAA) extension of another attendant
- VIP Wakeup Reminder Call — A VIP Wakeup reminder call.
- Miscellaneous Call — All other calls.

Call-Type Ordering Within Priority Levels?

If you use call-type ordering, calls to the attendant are first grouped by the queue priority level, then by call type, and, finally, in the order received.

Valid entries	Usage
y	Enter **y** if you want to present calls by call type. You can assign a type-disp button on the Attendant Console screen so that the attendant can review the call type for the active call.
n	Enter **n** if you wish the calls to be queued in chronological order by queue priority level.

The call types, in descending order of priority, are:

- Type 1 call: outgoing public-network calls receive answer supervision when the Answer Supervision Timer of the trunk group expires, even if the trunk is actually still ringing. Also, incoming calls when answered by the attendant.
- Type 2 call: incoming external public-network calls before they receive answer supervision or before the Answer Supervision Timer of the trunk group expires
- Type 3 call: all other calls (internal calls, conference calls, and tie-trunk calls of any type)

Note that external public-network calls have priority over all other calls including conference calls. And, answered public-network calls have priority over those calls not yet answered.

Field descriptions for page 4

```
change console-parameters                                              Page 4 of 4
                              CONSOLE PARAMETERS

  ASSIGNED MEMBERS ( Installed attendant consoles )
      Type       Grp  TN       Type       Grp  TN       Type       Grp  TN
   1:              1   1   11:              1   1   21:              1   1
   2:              1   1   12:              1   1   22:              1   1
   3:              1   1   13:              1   1   23:              1   1
   4:              1   1   14:              1   1   24:              1   1
   5:              1   1   15:              1   1   25:              1   1
   6:              1   1   16:              1   1   26:              1   1
   7:              1   1   17:              1   1   27:              1   1
   8:              1   1   18:              1   1   28:              1   1
   9:              1   1   19:              1   1
  10:              1   1   20:              1   1
```

Screen 69. Console Parameters — Default Attendant Group

ASSIGNED MEMBERS (Installed attendant consoles)

Display-only field that shows all attendants in the group. You administer the individual attendant consoles on the Attendant Console screen.

Grp

Display-only field that lists the Attendant Group number.

TN

Display-only field that lists the Tenant Partition number.

Coverage Answer Group

This screen establishes Call Coverage Answer Groups.

An answer group contains up to eight members who act as a coverage point for another user. For example, if several secretaries are responsible for answering a department's redirected calls, all the secretaries could be assigned to an answer group. The answer group is assigned a group number, and that group number appears in the department's coverage path. All phones in an answer group ring (alert) simultaneously. Any member of the group can answer the call.

Each coverage answer group is identified by a number from 1 through the maximum number allowed by your system configuration (refer to *Avaya MultiVantage™ Hardware Solutions Guide*). The members of the group are identified by their extension number. Any phone, including those administered without hardware (X-ported (but not attendants) can be assigned to a coverage answer group. Note that members whose extensions are X-ported will not be alerted.

Field descriptions for page 1

```
change coverage answer-group 3                                    Page 1 of 1
                        COVERAGE ANSWER GROUP

                      Group Number: 3___
                      Group Name: COVERAGE_GROUP_

GROUP MEMBER ASSIGNMENTS
   Ext      Name (first 26 characters)     Ext      Name (first 26 characters)
1: ________ ___________________________ 5: ________ __________________________
2: ________ ___________________________ 6: ________ __________________________
3: ________ ___________________________ 7: ________ __________________________
4: ________ ___________________________ 8: ________ __________________________
```

Screen 70. Coverage Answer Group

Group Number

A display-only field when the screen is accessed using an administration command such as **add** or **change**.

Group Name

Enter the group name you want to use to identify this group.

Tip:

Enter the extension numbers that are group members. This allows a list coverage answer group command to be used to list the phones that will be alerted. The list command can be used in conjunction with the list station, list coverage path, and list hunt group commands to determine stations involved in call coverage. This makes it possible to follow call coverage for any extension, allowing the administrator to easily track call coverage paths.

Valid entries	Usage
Up to 27 characters	For example, typing pool, room 12, secy, and so on.

Ext

Enter the extension number (may not be a Vector Directory Number extension) for each member of this coverage answer group.

Valid entries	Usage
An assigned extension for a station.	

Name

This display-only field indicates the name assigned when the member’s phone is administered.

Coverage Path

This screen implements Call Coverage Paths. The screen provides the means to specify the call coverage criteria, the points in the coverage path used to redirect calls, and the number of times a principal's phone rings before the call redirects to coverage.

Field descriptions for page 1

```
change coverage path 2                                             Page  1 of  1
                          COVERAGE PATH

            Coverage Path Number: 2
                                        Hunt After Coverage: n
                Next Path Number: ___   Linkage: ___  ___

COVERAGE CRITERIA

  Station/Group Status   Inside Call   Outside Call
            Active?           n             n
              Busy?           y             y
      Don't Answer?           y             y Number of Rings:2
               All?           n             n
DND/SAC/Goto Cover?           y             y

COVERAGE POINTS

   Terminate to Coverage Pts. with Bridged Appearance? n

     Point1: _________ Point2: _________ Point3: _________
     Point4: _________ Point5: _________ Point6: _________
```

Screen 71. Coverage Path screen

Coverage Path Number

A display-only field indicating the coverage path being administered.

Hunt After Coverage

Valid entries	Usage
y	Coverage treatment continues by searching for an available station in a hunt chain that begins with the hunt-to-station assigned on the station screen of the last coverage point.
n	Coverage treatment is terminated; the call is left at the last available location (principal or coverage point).

Next Path Number

Enter the next coverage path in a coverage path chain. Refer to ‘‘Call Coverage’’ on page 1416 for more information. If the coverage criteria of the current coverage path is not satisfied, the system steps down this chain until it finds a coverage path with redirection criteria that matches the call status. If the chain is exhausted before the system finds a match, the call does not redirect to coverage. No path number here indicates that this path is the only path for the principal.

Valid entries	Usage
1 to **999**	

Linkage

Display-only fields that show the (up to) two additional coverage paths in the coverage path chain. (See above.)

COVERAGE CRITERIA

COVERAGE CRITERIA are the conditions that, when met, cause the call to redirect to coverage. Assign one of the following:

Valid entries	Usage
Active	Calls redirect if at least one call appearance is busy.
Busy	Calls redirect if all call appearances that accept incoming calls are busy.
Don’t Answer	Calls redirect when the specified number of rings has been exceeded.
All	Calls redirect immediately to coverage and overrides any other criteria with a **y** in this column.
DND/SAC/Goto Cover	Must be assigned before a user can activate Do Not Disturb (Hospitality Services), Send All Calls (SAC), or Go to Cover features. Allows a calling user, when calling to another internal extension, to redirect a call immediately to coverage by pressing a GO TO COVER button. Allows a principal temporarily to direct all incoming calls to coverage, regardless of the other assigned coverage criteria by pressing the SEND ALL CALLS (or DO NOT DISTURB) button. Send All Calls also allows covering users to temporarily remove their phones from the coverage path.

Number of Rings

Enter the number of rings.

Valid entries	Usage
1 through **99**	This is the number of rings a user's phone rings before the system redirects the call to the first point in the coverage path.

COVERAGE POINTS

Terminate to Coverage Pts. with Bridged Appearances

Valid entries	Usage
y	Allows a call to alert as both a bridged call and a redirected call.
n	The call skips the coverage point if it has already alerted as a bridged call.

Point1, Point2, Point3, Point4, Point5, Point6

The alternate destinations that comprise a coverage path. Coverage points must be assigned sequentially beginning with Point 1 (do not leave gaps). Each path can have up to six coverage points.

Valid entries	Usage
extension	Redirects the call to an internal extension or announcement
attd	Redirects the call to the attendant or attendant group. If the system has Centralized Attendant Service (CAS), the call goes to the CAS attendant.
h1 to **h255**	Redirects the call to the corresponding hunt-group. For example, enter "**h32**" if you want a coverage point routed to hunt group 32. (Refer to ''Hunt Group'' on page 840 for more information.)
c1 to **c750**	Redirects the call to the corresponding coverage answer group. For example, enter "**c20**" if you want a coverage point routed to call coverage answer group 20. (Refer to ''Coverage Answer Group'' on page 670 for more information.)

Valid entries	Usage
r1 to **r999**	Redirects the call to the corresponding remote coverage point number. For example, enter "**r27**" if you want a coverage point routed to remote coverage point 27. (Refer to ''Remote Call Coverage Table'' on page 1037 for more information.)
v + extension	Redirects the call to the corresponding VDN extension. For example, enter "**v12345**" if you want the last administered coverage point to be the VDN associated with extension 12345. Note that a Vector Directory Number may be used only as the last administered point in a coverage path.

If calls redirect to an AUDIX in a DCS network, administer a unique Hunt Group screen. Assign the AUDIX extension in the Group Extension field. If the AUDIX is connected to the local node, set the Message Center field to **audix**; if the AUDIX is connected to another node, set the Message Center field to **rem-audix**.

If calls redirect to Message Center (a special Uniform Call Distribution hunt group), AUDIX, or to the attendant, do not list any subsequent coverage points. These calls will normally queue and never redirect to another coverage point. Calls to any hunt group will queue if possible. Calls redirect from a hunt group only if all hunt group members are busy and either the queue is full or there is no queue.

If the Coverage of Calls Redirected Off-Net feature is not enabled, a remote coverage point will function as the last point in the coverage path, because the system will no longer have control of the call once it has redirected off-net. However, if the Coverage of Calls Redirected Off-Net feature is enabled, a call redirected off-net can be monitored by the system and brought back for further call coverage processing.

Crisis Alert System Parameters

This screen allows you to define the system parameters associated with sending crisis alert messages.

Field descriptions

```
change system-parameters crisis-alert
                          CRISIS ALERT SYSTEM PARAMETERS

ALERT STATION
    Every User Responds? n

ALERT PAGER
            Alert Pager? y
  Originating Extension: 7768
      Crisis Alert Code: 911
                Retries: 5
   Retry Interval (sec): 30
            Main Number: 303-555-0800

                      Pager Number          Pin Number
                      1: 3035559001         1: 7614567890
                      2: 123456789012345    2: ppp1234567890pp
                      3: 123456789012345    3: ppp1234567890pp

                  DTMF Duration - Tone (msec): 100   Pause (msec): 100
```

Screen 72. Crisis Alert System Parameters screen

Field description

Every User Responds

Controls who needs to respond to a crisis alert.

Valid entries	Usage
y	If set to **y,** all users who have a crisis alert button are notified and must clear the alert for every emergency alert. Assign crisis alert buttons only to attendant consoles and stations that must be notified of an emergency call.
n	If set to **n**, all users are notified, but only one user needs to acknowledge an alert. This user may be the attendant or any other digital telephone with a crisis alert button. When the alert is acknowledged by one user, the alert is cleared at all stations except the one that acknowledged the alert.

Alert Pager

Valid entries	Usage
y/n	Enter ***y*** to use Crisis Alert to a Digital Pager.

Originating Extension

Used as the extension originating the call to send a crisis alert message to a pager. Displays when the Alert Pager field is **y**. This field requires an entry before submitting the screen.

Valid entries	Usage
1 - 7 digits	Requires a valid unassigned extension according to the dial plan.

Crisis Alert Code

Displays when the Alert Pager field is **y**. This field requires an entry before submitting the screen.

Valid entries	Usage
1 - 3 digits	The numbers in this field are the first 3 digits in the crisis alert pager message. Avaya recommends you enter the numbers used to call the local emergency service or any digits used for an emergency situation (for example, 911).

Retries

Displays when the Alert Pager field is **y**.

Valid entries	Usage
0 - 10	The number of times the system tries to send out the alert message in case of an unsuccessful attempt. This increases the chances that the pager receives a crisis alert message.

Retry Interval (sec)

Displays when the Alert Pager field is **y**. This field is not used unless the Retries field is **1-10**.

Valid entries	Usage
30 - 60	The administrable time period (in seconds) between retries. If an attempt to call the pager fails, the retry call attempts after the retry interval period.

Main Number

The main phone number to the location or a location code. This field is optional and does not require an entry. Displays when the Alert Pager field is **y**.

Valid entries	Usage
digits **0-9** **-** (dash)	Enter a number up to 15 digits to identify the location where the crisis alert call originated. It can be the main number to the location or a numerical identification. Any dashes are for display purposes only and not included in the message sent to the pager. This entry is the last group of digits displayed in the pager message.

Pager Number

Displays when the Alert Pager field is **y**. One of these fields must have a number or the screen cannot be submitted.

Valid entries	Usage
1 to 15 digits	Any dashes are for display purposes only and not included in the message sent to the pager One of the pager number fields must have a number or the screen cannot be submitted.

Pin Number

This field can be used for any combination of the pager pin number and pauses or left blank. Displays when the Alert Pager field is **y**.

Valid entries	Usage
digits **0-9** **p**(ause) **#** (pound) ***** (star)	Enter a number up to 15 digits. A pause (about 2 seconds) is for timing of the message. For instance, after the pin number you may want to have a pause to allow time for the pager service to set up the correct pager message box. If the pager service requires you to submit a PIN number, enter it here.

DTMF Duration - Tone (msec)

The length of time the Dual-Tone Multi-Frequency (DTMF) tone is heard for each digit. Displays when the Alert Pager field is **y**.

Valid entries	Usage
20 - 2550	Enter a number in increments of 10.

Pause (msec)

The length of time between DTMF tones for each digit. Displays when the Alert Pager field is **y**.

Valid entries	Usage
20 - 2550	Enter a number in increments of 10.

Data modules

The following section provides descriptions of standard fields on Data Module screens. Some of the fields are used for specific data module types; others are used for all data modules. Unique fields and fields requiring special consideration are listed with the appropriate data module descriptions in this book.

Field descriptions for page 1

```
change data-module 30                                              Page   1 of 2
                                DATA MODULE

   Data Extension: 30            Name: 27                          BCC:
             Type: data-line___   COS: 1
             Port: _______        COR: 1
              ITC: restricted__    TN: 1                  Connected to: dte


  ABBREVIATED DIALING
    List1:

  SPECIAL DIALING OPTION:



  ASSIGNED MEMBER (Station with a data extension button for this data module)

          Ext     Name
      1: 1002    27 character   station name
```

Screen 73. Data Module

Data Extension

Enter the extension assigned to the data module.

Valid entries	Usage
1 - 7-digit number	Must agree with the dial plan.

Name

Enter the name of the user associated with the data module. The name is optional and can be blank.

Valid entries	Usage
Up to 27 alphanumeric characters	

BCC

(Bearer Capability Class) Used with Data Line, Netcon, Processor Interface, Processor/Trunk, and System Port Data Modules. Appears when the ISDN-PRI or ISDN-BRI Trunks field is **y** on the System-Parameters Customer-Options screen**.** The value in this field corresponds to the speed setting of the data module. This field may be compared with the BCC value in an associated routing pattern when attempted calls utilizing the data module fail to complete. The BCC values must be the same.

Refer to ‘‘Generalized route selection’’ on page 1567 for a detailed description of Bearer Capability Classes (BCC) and their ability to provide specialized routing for various types of voice and data calls. The BCC value is used to determine compatibility when non-ISDN-PRI facilities are connected to ISDN facilities (ISDN-PRI Interworking).

Valid entries	Usage
1	Relates to 56-bkps
2, **3**, **4**	Relates to 64 kbps

Type

Enter the type of data module.

Valid entries	Usage
7500	Assigns a 7500 Data Module. The 7500 data module supports automatic TEI, B-channel, maintenance and management messaging, and SPID initialization capabilities. BRI endpoints, both voice and/or data, are assigned to either the ISDN-BRI - 4-wire S/T-NT Interface circuit pack or the ISDN-BRI - 2-wire U circuit pack. Each can support up to 12 ports. Since BRI provides multipoint capability, more than one ISDN endpoint (voice or data) can be administered on one port. For BRI, multipoint administration allows for telephones having SPID initialization capabilities, and can only be allowed if no endpoint administered on the same port is a fixed tie endpoint and no station on the same port has B-channel data capability. Currently, multipoint is restricted to 2 endpoints per port.
announcement	Assigns an announcement data module. The announcement data module is built-in to the integrated announcement circuit pack and is administered using the Announcement Data Module screen. This data module (in conjunction with an administered Netcon Data Module in DEFINITY Server SI configurations) allows the system to save and restore the recorded announcements file between the announcement circuit pack and the system memory.

Valid entries	Usage
data-line	Assigns a Data Line Data Module. The Data Line Data Module (DLDM) screen assigns ports on the Data Line circuit pack (DLC) that allows EIA 232C devices to connect to the system. The DLC, with a companion Asynchronous Data Unit (ADU), provides a less expensive data interface to the system than other asynchronous DCP data modules. The DLC supports asynchronous transmissions at speeds of Low and 300, 1200, 2400, 4800, 9600, and 19200 bps over 2-pair (full-duplex) lines. These lines can have different lengths, depending on the transmission speed and wire gauge. The DLC has 8 ports. The connection from the port to the EIA device is *direct*, meaning that no multiplexing is involved. A single port of the DLC is equivalent in functionality to a data module and a digital line port. The DLC appears as a data module to the Digital Terminal Equipment (DTE) and as a digital line port to the switch. The DLC connects the following EIA 232C equipment to the system: ■ Printers ■ Non-Intelligent Data Terminals ■ Intelligent Terminals, Personal Computers (PCs) ■ Host Computers ■ Information Systems Network (ISN), RS-232C Local Area Networks (LANs), or other data switches.
ethernet	Assigns an Ethernet data module. The Ethernet Data Module screen assigns the 10BaseT port on the Control-LAN (C-Lan) circuit pack. This port provides a TCP/IP connection to network hub or LAN. Refer to *Administration for Network Connectivity for Avaya MultiVantage*™ *Software* for more information on Ethernet data modules.

Valid entries	Usage
netcon	Assigns a Netcon Data Module. Netcon data modules are the Processor Data Modules (PDMs) that are integrated into the system’s network control ports that provide asynchronous circuit switched interfaces to the maintenance and administration terminals, Hospitality journal printers, and CDR digital output. They are characterized by their special locations, that is, special port identifications. **NOTE:** The Netcon data module is only applicable to DEFINITY Server SI configurations. For DEFINITY Server R and later configurations, use the **system-port** command.

Valid entries	Usage
pdm	Assigns a DCE interface for Processor/Trunk Data Modules. These screens assign Modular Processor Data Modules (MPDMs) and Modular Trunk Data Modules (MTDMs). One screen is required for assigning MPDMs (700D), 7400B, 7400D or 8400B Data Module, and another screen for MTDMs (700B, 700C, 700E, 7400A). One screen must be completed for each MPDM, 7400B, 7400D, 8400B or MTDM. The MPDM, 7400B, or 8400B Data Module provides a Data Communications Equipment (DCE) interface for connection to equipment such as data terminals, CDR output devices, on-premises administration terminal, Message Server, Property Management System (PMS), AUDIX, and host computers. It also provides a Digital Communications Protocol (DCP) interface to the digital switch. (DCE is the equipment on the network side of a communications link that provides all the functions required to make the binary serial data from the source or transmitter compatible with the communications channel.) The MTDM provides an Electronic Industries Association (EIA) Data Terminal Equipment (DTE) interface for connection to off-premises private line trunk facilities or a switched telecommunications network and a DCP interface for connection to the digital switch. (DTE is the equipment comprising the endpoints in a connection over a data circuit. For example, in a connection between a data terminal and a host computer, the terminal, the host, and their associated modems or data modules make up the DTE.) The MTDM or 7400A Data Module also can serve as part of a conversion resource for Combined Modem Pooling.
ppp	Assigns a Point-to-Point Protocol data module. The PPP Data Module screen assigns a synchronous TCP/IP port on the C-Lan. These ports are tailored to provide TCP/IP connections for use over telephone lines. Refer to *Administration for Network Connectivity for Avaya MultiVantage*™ *Software* for more information on Point-to-Point data modules.

Valid entries	Usage
procr-intf	Assigns a Processor Interface Data Module. The Processor Interface data modules are the Processor Data Modules (PDMs) that are integrated into the system's synchronous/asynchronous Processor Interface circuit pack ports. **NOTE:** The Processor Interface data module is applicable only to DEFINITY Server SI configurations. They are used to provide the following interfaces: ■ 3B/Call Management System (maximum of 1) ■ 3B/Message Server (maximum of 1) ■ Distributed Communications System (maximum of 8) ■ AUDIX (maximum of 1) **NOTE:** Not all maximums can be achieved at the same time Connections for these interfaces are achieved via a digital line port and MPDM combination, and/or in the case of DCS or ISDN-PRI, via a DS1 interface. One direct EIA connection is available (labeled as Processor Interface on the back of the Control Cabinet) for simplex operation. When used, the physical channel assignment (see below) must be "01". Use of the EIA connection eliminates the need for one digital line port/MPDM combination. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information on Processor Interface data modules.
system-port	Assigns a System Port Data Module.
tdm	Assigns a DTE interface for Processor/Trunk Data Modules. Refer to the pdm entry above.
wcbri	Assigns a World Class BRI Data Module.

Valid entries	Usage
x.25	Assigns an X.25 Data Module in DEFINITY Server MultiVantageR configurations for communications to Adjuncts and other nodes in a DCS network. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information on X.25 data modules.

COS

Enter the desired class of service.

Valid entries	Usage
0 - 15	Select the allowed features.

Remote Loop-Around Test

Used with Processor/Trunk and X.25 Data Modules. Appears when the Type field is **pdm**, **tdm**, or **x.25**.

Valid entries	Usage
y/n	For Processor/Trunk Data Modules, enter **y** if the data module supports a loop-back test at the EIA interface. In general, Avaya equipment supports this test but it is not required by Level 2 Digital Communications Protocol. Enter **n** to abort a request for this test.
y/n	For X.25 Data Modules, enter **y** to allow remote loop-around tests on this port. Refer to the *Maintenance for Avaya™ MultiVantage and DEFINITY® Server* manual for more information about remote loop-around tests. Enter **n** to abort a request for this test.

Maintenance Extension

Used with Netcon and Processor Interface Data Modules.

Valid entries	Usage
Enter the extension number required to perform maintenance functions on the standby netcon physical channel in a duplicated system.	The standby remote loop around tests fails if this field is not administered.

Physical Channel

Used with Netcon and Processor Interface Data Modules. The Physical Channel number is referred to on associated system forms as the Interface Link number.

Valid entries	Usage
01 - 08	For Processor Interface Data Modules, enter the 2-digit circuit number of the Processor Interface port. A multi-carrier cabinet system supports the use of two Processor Interface circuit packs, the first circuit pack (mounted in Control Carrier A) supports physical channels or links 01 through 04; the second (mounted in Control Carrier A) supports physical channels or links 05 through 08. A single-carrier cabinet system supports one Processor Interface circuit pack and physical channels or links 01 through 04 only.
01 - 04	For DEFINITY Server CSI configurations. For Netcon Data Modules, enter a netcon data channel.
01-08	For DEFINITY Server SI configurations. For Netcon Data Modules, enter a netcon data channel.

Board

Used with Announcement Data Modules. Enter the five character announcement circuit pack number that identifies the physical circuit pack to which the announcement module is connected. You can enter **X** in this field to indicate that there is no hardware associated with this port assignment.

The five character announcement board number is comprised of:

Characters	Meaning	Value
1-2	Cabinet Number	01 through 44 (DEFINITY Server R configurations)
		01 through 03 (DEFINITY Server SI configurations)
3	Carrier	A through E
4-5	Slot Number or X	1 through 20

Port

Used with 7500, Data Line, Ethernet, Processor/Trunk, PPP, System Port, X.25, and World Class BRI Data Modules. Specifies a port location to which the data module is connected.

Characters	Meaning	Value
1-2	Cabinet Number	01 through 44 (DEFINITY Server R configurations) 01 through 03 (DEFINITY Server SI configurations)
3	Carrier	A through E
4-5	Slot Number	0 through 20
6-7	Circuit Number	01 through 04 (x.25 circuit pack) 01 through 31 (DEFINITY Server SI configurations)

NOTE:
You can enter X in the Port field to indicate that there is no hardware associated with the port assignment (also known as administration without hardware). These stations are referred to as “phantom stations.” If this data module is designated as a secondary data module (Secondary data module set to **y**) X cannot be entered into this field. The port of a primary data module cannot be changed to X if a secondary data module is administered.

COR

Enter the desired class of restriction.

Valid entries	Usage
0 - 95	Select the allowed restriction.

Multimedia

Used with the 7500 and World Class BRI Data Modules. Appears only if, on the System-Parameters Customer-Options screen, the MM field is **y**.

Valid entries	Usage
y/n	Enter **y** to make this data module part of a multimedia complex.

Destination Number

Used with X.25 and Processor Interface data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

MM Complex Voice Ext

Used with 7500 and World Class BRI Data Modules. This field contains the number of the associated phone in the multimedia complex. This field appears only after you set the Multimedia field to **y**. This field is left blank until you enter the data module extension in MM Complex Data Ext on the Station screen.

Valid entries	Usage
Valid values conform to your dial plan	Once you complete the field on the station screen, these two extensions are associated as two parts of a one-number complex, which is the extension of the telephone.

Secondary data module

Used with Processor/Trunk Data Modules. Appears only when the Type field is **pdm**. The primary data module must be administered before the secondary data module may be added. If the Port field is **X**, the Secondary data module field cannot be **y**.

Valid entries	Usage
y	This PDM is the secondary data module used for Dual I-channel AUDIX networking.
n	This is the primary PDM, or if this data module is not used for AUDIX networking.

Baud Rate

Used with X.25 Data Modules. The maximum raw data transmission speed.

Valid entries	Usage
300 **1200** **2400** **4800** **9600** **19200**	
switched	You can enter this if the Cable Type field is **none** on the PGATE screen.

ITC

(Information Transfer Capability) Used with 7500, Announcement, Netcon, Processor/Trunk, and Processor Interface Data Modules. Appears only when the Comm Type field is **56k-data** or **64k-data**. Indicates type of transmission facilities to be used for ISDN calls originated from this endpoint. Does not display for voice-only or BRI stations.

Valid entries	Usage
restricted	Either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission (that is, a sequence of 8 digital zeros are converted to a sequence of 7 zeros and a digital 1).
unrestricted	Only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission (that is, digital information is sent exactly as is).

TN

Valid entries	Usage
1 - 20	Enter the Tenant Partition number.

Establish Connection

Used with X.25, Point-to-Point, and Processor Interface data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

Connected to

Used with Data Line and Processor/Trunk (**pdm** selection) Data Module. This field shows to what the Asynchronous Data Unit (ADU) is connected.

Valid entries	Usage
dte	Data Terminal Equipment. Used with Data Line and Processor/Trunk Data Modules.
isn	Information Systems Network. Used with Data Line and Processor/Trunk Data Modules.

Connected Data Module

Used with X.25 and Processor Interface data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

Enable Link

Used with X.25, Ethernet, Point-to-Point, and Processor Interface data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information. This field is in different locations on the screen for different data module types.

Node Name

Used with Ethernet and Point-to-Point data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

Subnet Mask

Used with Ethernet data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

Broadcast Address

Used with Ethernet data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

Network uses 1's for Broadcast Addresses

Used with Ethernet data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

IP Address Negotiation

Used with Point-to-Point data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

PDATA Port

Used with System Port Data Modules. Enter a seven-digit alphanumeric port location to which the data module is connected. Used to relate the physical PDATA port to which the mode 3 portion of the system port is connected. This entry must be assigned to a port on a PDATA Line Board.

Valid entries	Usage
01 through **22**	First and second characters are the cabinet number
A through **E**	Third character is the carrier
01 through **20**	Fourth and fifth characters are the slot number in the carrier
01 through **12**	Sixth and seventh characters are the circuit number

Endpoint Type

Used with X.25 Data Modules. An endpoint type is a type of packet switched data endpoint that uses X.25 call control procedures. The X.25 Endpoint connects to external ports on the PGATE board and to the TDM bus via a DS1 trunk. Ports connected to the adjunct endpoint can be either DTEs or DCEs. The type of endpoint (DTE or DCE) is administrable on the “data-mod” screen.

Valid entries	Usage
adjunct	Mandatory entry

Link

Used with X.25, Ethernet, Point-to-Point, and Processor Interface data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information. This field is in different locations on the screen for different data module types.

DTE/DCE

Used with X.25 Data Modules. Specifies how the above endpoint type acts.

Valid entries	Usage
dte	Data Terminal Equipment
dce	Data Communications Equipment

Error Logging

Used with X.25 Data Modules.

Valid entries	Usage
y/n	Enter **y** to record X.25 protocol errors in the hardware error log. Refer to the *Maintenance for Avaya™ MultiVantage and DEFINITY® Server* for more information about error logs.

Permanent Virtual Circuit

Used with X.25 Data Modules.

Valid entries	Usage
y/n	Enter **y** to indicate that the logical channels for PVC are allowed for this port. Cannot be changed.

Highest PVC Logical Channel

Used with X.25 Data Modules. Specifies how the above endpoint type acts.

Valid entries	Usage
1 - 64	Indicates that 1 to 64 Logical Channels are allowed on this port. Cannot be changed.

Switched Virtual Circuit

Used with X.25 Data Modules.

Valid entries	Usage
y/n	Enter **y** to indicate that the Switched Virtual Circuit is not allowed for this port. Cannot be changed.

Digits

Used with Point-to-Point data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

Node Name

Used with Point-to-Point data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

CHAP

Used with Point-to-Point data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

CHAP Secret

Used with Point-to-Point data modules. Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

Abbreviated Dialing List1

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Supports Data Hot Line. This field can be left blank.

Valid entries	Usage
e	Enhanced
g	Group. You also must enter a group list number.
p	Personal. You also must enter a personal list number.
s	System.

Special Dialing Option

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Identifies the type of dialing for calls when this data module originates calls.

Valid entries	Usage
hot-line	
default	
blank	For regular (normal) keyboard dialing.

Abbreviated Dialing Dial Code

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Appears only when the Special Dialing Option field is **default**. Enter a list number associated with the AD list. When the user goes off-hook and enters a carriage return following the DIAL prompt, the system dials the AD number. The data call originator can also perform data-terminal dialing by specifying a dial string that may or may not contain alphanumeric names.

Valid entries	Usage
0-999	Enter dial code within range of abbreviated dialing list type.

Hot Line Destination — Abbreviated Dialing Dial Code

Used with 7500, Data Line, Netcon, Processor/Trunk, Processor Interface, and World Class BRI Data Modules. Appears only when the Special Dialing Option field is **hot-line**. Entry in this field supports Data Hot Line.

Valid entries	Usage
0-999	This number is associated with the AD List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.

Default Dialing Abbreviated Dialing Dial Code

Only appears when the Special Dialing Option field is **default**. When the user goes off-hook and enters a carriage return following the DIAL prompt, the system dials the AD number. The data call originator can also perform data-terminal dialing by specifying a dial string that may or may not contain alphanumeric names.

Valid entries	Usage
0-999	Enter a list number associated with the abbreviated dialing list.

CIRCUIT SWITCHED DATA ATTRIBUTES

Used with 7500 and World Class BRI Data Modules.

NOTE:
These fields represent defaults needed for modem pooling conversion resource insertion when the endpoint does not support data query capability and administered connections. These fields have no significance for data modules providing data query [all Avaya -supported ISDN-BRI data modules (7500 and ADM)]. For Avaya ISDN-BRI or World Class ISDN-BRI data modules, use the default settings.

Default Duplex

Used with 7500 and World Class BRI Data Modules. Used to identify the duplex mode.

Valid entries	Usage
full	Allows simultaneous two-way transmission.
half	Allows only one transmission direction at a time.

Default Mode

Used with 7500 and World Class BRI Data Modules. Used to identify the data mode.

Valid entries	Usage
sync	Synchronous
async	Asynchronous

Default Speed

Used with 7500 and World Class BRI Data Modules. Used to identify the data rate.

Valid entries	Usage
1200 **2400** **4800** **19200**	
56000 **64000**	Can be entered when the Default Mode field is **sync**.

ASSIGNED MEMBER

Ext and Name

Used with Data Line, Announcement, Netcon, Processor/Trunk, Processor Interface, and System Port Data Modules. Appears the extension number and name of the user (previously administered) with associated Data Extension buttons who shares the module.

DATA MODULE CAPABILITIES

Default ITC

Used with 7500 and World Class BRI Data Modules.

Valid entries	Usage
restricted	For a WCBRI endpoint used as an administered connection.

Default Data Applications

Used with 7500 and World Class BRI Data Modules. Used to identify mode to be Used with originating data calls when the mode is not specified with the calling parameters. This mode is also used for terminating trunk calls that do not have bearer capability specified or administered connections. Refer to ''Generalized route selection'' on page 1567 for additional information.

Valid entries	Usage
M0	Mode 0. Use this setting for a WCBRI endpoint used as an administered connection.
M1	Mode 1
M2_A	Mode 2 asynchronous
M2_S	Mode 2 synchronous
M3/2	Mode 3/2 adaptable
blank	

MM Complex Voice Ext:

Used with 7500 and World Class BRI Data Modules. This display-only field contains the number of the associated phone in the multimedia complex. It only appears when the Multimedia field is **y**. This field is left blank until you enter the data module extension in MM Complex Data Ext field on the Station screen. Once you complete the field on the Station screen, these two extensions are associated as two parts of a one-number complex, which is the extension of the phone. Valid values conform to your dial plan.

Field descriptions for page 2

```
change data-module 30                  DATA MODULE                  Page 2 of 2
        CAPABILITIES
                       KYBD Dialing? y                            Configuration? n
                           Busy Out? n
        SPEEDS
              Low? y            1200? y            4800? y            19200? y
              300? y            2400? y            9600? y       Autoadjust? n
        OPTIONS
                     Permit Mismatch? n                           Dial Echoing? y
                 Disconnect Sequence: two-breaks                   Answer Text? y
                              Parity: even           Connected Indication? y
```

Screen 74. Data Line Data Module — if KYBD Dialing is y

```
change data-module 30                       DATA MODULE                    Page 2 of 2
        CAPABILITIES
                       KYBD Dialing? n
                           Busy Out? n
        SPEEDS
              Low? y             1200? y             4800? y            19200? y
              300? y             2400? y             9600? y
        OPTIONS
                    Permit Mismatch? n
```

Screen 75. Data Line Data Module — if KYBD Dialing is n

Refer to ‘‘DLC Option Settings’’ on page 703 for additional information when assigning entries for the remaining fields on the screen.

KYBD Dialing

This option must be enabled to allow data endpoints to originate calls via the EIA 232C interface and obtain ASCII feedback text. When enabled, the user gets the dial prompt. This option normally is enabled for “originate/receive” DTE that has a need to set up data calls. If this option is disabled, originations cannot be done at the DTE and text feedback does not occur at the DTE during call setup/take down. Data call answering is still allowed but without text feedback.

NOTE:
ADU-type hunt groups connecting the system to terminal servers on a host computer should have these hunt group extensions assigned as “no” keyboard dialing.

Valid entries	Usage
y/n	Enter **y** to allow keyboard dialing. This enables the data endpoint to receive and transmit text during call origination or termination. Low must be **n**.

Configuration

Appears when the KYBD Dialing field is **y**. This option normally is enabled for "originate/receive" DTE such as non-intelligent terminals and disabled for intelligent devices such as computers. The KYBD Dialing field must be **y** with this option.

Valid entries	Usage
y/n	Enter **y** to allow the viewing and changing of options from the DTE.

Busy Out

This option should be enabled for DTEs that are members of a hunt group and to allow "busy out" when DTE turns power off so that calls do not terminate on that DTE.

Valid entries	Usage
y/n	Enter **y** to place the DLC port in a busied-out state once the DTE control lead to the DLC is dropped.

SPEEDS

Enter **y** to select operating speeds as follows:

Valid entries	Usage
Low	Enter **y** to instruct the DLC to operate at a low speed from 0 to 1800 bits per second (bps). Enter **n** if the KYBD Dialing field is **y**.
300, **1200**, **2400**, **4800**, **9600**, or **19200**	Enter **y** beside the desired operating speed. Enter **n** if the speed is not desired. The DLC can be any one of these speeds. The speed is matched for the duration of the call, from call setup to call takedown. When multiple speeds are selected (select three or more, do not select just two speeds) and autoadjust is disabled, the DTE's speed must be the highest selected speed. This is required because all feedback text is delivered to the DTE at the highest selected speed.
Autoadjust	Appears when the KYBD Dialing field is **y**. Enter **y** which tells the DLC port to automatically adjust to the operating speed and parity of the DTE it is connected to. Enter **n** if this option is not desired. Autoadjust can be selected with any of the speeds selected in the previous step. Autoadjust allows the DLC port to determine the speed and parity of the DTE and then match itself to this speed. Autoadjust only applies to calls originated by the user through Keyboard Dialing.

Permit Mismatch

This option allows the EIA interface to operate at a rate different than that agreed to in the data module handshake. (The data module handshake is always the highest compatible rate as determined by the reported speed option of each data module.) Permit Mismatch eliminates the need to change the DTE/DLC speed every time a call is placed to/from an endpoint operating at a different speed. When this option is enabled, the DLC reports the highest optioned speed and all the lower speeds (or the previously selected autoadjust speed) during the handshake process.

Valid entries	Usage
y/n	Enter **y** to instruct the DLC to operate at the highest selected speed, which is a higher rate than the far-end data module.

CAUTION:

Caution must be used when using this option to send information from a DTE/DCE that is transmitting data at higher rates than that of the far end. Sustained usage of this type transmission results in loss of data. Whenever this option is enabled, the DTE must match the highest speed selected for the associated DLC port.

This option is intended to be used by a DTE device operating locally at a higher baud rate than that of its far-end connection but transmitting relatively low amounts of data (for example, a user typing at a terminal). Also, this option may be selected whether Keyboard Dialing is selected.

NOTE:

The Low speed setting is not reported as an available speed when the Permit Mismatch field is **y**.

Dial Echoing

Appears when the KYBD Dialing field is **y**.

Valid entries	Usage
y/n	Enter **y** to echo characters back to the DTE. Dial echoing should be disabled when keyboard dialing is done by an intelligent device.

Disconnect Sequence

Appears when the KYBD Dialing field is **y**. Selects the sequence for a disconnect.

Valid entries	Usage
long-break	A long-break is greater than 2 seconds.
two-breaks	Two-breaks is within 1 second.

Answer Text

Appears when the KYBD Dialing field is **y**. This option enables text feedback that is normally delivered to the DTE when a call is answered or disconnected. The Answer Text option applies to DLC-generated text as well as text received from the system. If this option is disabled, the system still generates the text, but the DLC prevents it from being sent to the device.

This applies to the following messages:

— INCOMING CALL

— ANSWERED

— DISCONNECTED

— DISCONNECTED OTHER END

This option usually is disabled when the answering DTE is a computer or an intelligent device.

Valid entries	Usage
y/n	Enter **y** to allow text messages to be delivered to the DTE when a call is being answered.

Parity

Appears when the KYBD Dialing field is **y**. Select the desired type of parity. The DLC generates the parities when call setup text is sent to the DTE. The DLC does not check the parity when receiving dialing characters. Parity has nothing to do with the far end; it is used by the DLC to terminal communications during call setup. Set to match the connected DTE.

Valid entries	Usage
even	
odd	
mark	
space	

Connected Indication

Appears when the KYBD Dialing field is **y**. This option generates a "CONNECTED" message to the DTE when the connection has been established. If the KYBD Dialing field is **n**, the connected indication is provided by the DLC activating its EIA 232C control lead.

Valid entries	Usage
y/n	Enter **y** to select this option.

DLC Option Settings

The following provides additional information on the option settings for DLCs when used with the following types of devices:

- Printers
- Non-intelligent terminals
- Data terminals and personal computers
- Host computers
- Information Systems Network (ISN)

Printers

A DLC port with a companion ADU, when attached to a printer, usually terminates a data call. Therefore, in this connection, the printer is the endpoint device. The originating device may be attached to a DCP mode 2 data module (such as the MPDM) or the DLC. A Z3A ADU extends the range of the EIA 232C connection.

When a receive-only printer (or any printer that does not generate the Transmit Data and DTR leads) is used, the ADU must be powered from a small plug-mounted transformer (2012D, or equivalent) connected to pins 7 and 8 of the modular jack. (Refer to *ADU User Manual* for details.)

An ADU cannot be used if the printer has hardware flow control using the Clear To Send (CTS) lead. An ADU can be used, however, if the printer is using software flow control.

A printer connected to a DLC is usually assigned as a line. Table 15 lists the option settings for printer connections.

Table 15. DLDM screen settings for printer connection

Field on screen	Option	Comments
Speed	Highest speed at which the Printer operates	Subject to distance limitations; Autoadjust not used
KYBD Dialing	no	
Busy Out	yes	If printer is member of Hunt Group
Permit Mismatch	yes	No, if printer is low speed
Parity	-	Don’t care
Dial Echoing	-	Don’t care
Disconnect Sequence	-	Don’t care
Answer Text	-	Don’t care
Connected Indication	-	Don’t care
Configuration	no	

Non-intelligent terminals

A non-intelligent terminal connected to the DLC usually is assigned as a line. Table 16 lists the option settings for non-intelligent terminals.

Table 16. DLDM screen settings for connection to non-intelligent terminals

Field On screen	Option	Comments
Speed	All speeds at which the terminal can operate; autoadjust	Subject to distance limitations; Autoadjust when the KYBD Dialing field is **y** and the Terminal can generate an ASCII “return”
KYBD Dialing	yes	
Busy Out	no	Yes, if terminal is member of a hunt group
Permit Mismatch	yes	-

Continued on next page

Table 16. DLDM screen settings for connection to non-intelligent terminals (Continued)

Field On screen	Option	Comments
Parity	Same as DTE	
Dial Echoing	yes	Only if the KYBD Dialing field is **y**
Disconnect Sequence	2	Depends on terminal
Answer Text	yes	
Connected Indication	-	Don’t care
Configuration	yes	

Data terminals and personal computers

An intelligent data terminal or a personal computer (PC) attached to a DLC can either originate or terminate a data call. A single ADU at the site of the originating device extends the distance signals can travel to the switch (the model ADU depends on the terminal connector). An analog telephone can be attached to this arrangement whenever an ADU uses the standard building wiring. Table 17 lists the option settings used for data terminal and personal computer connections.

Table 17. DLDM screen settings for connection to data terminal or personal computer

Field on screen	Option	Comments
Speed	All speeds at which the Data Terminal or PC can operate	Subject to distance limitations; Autoadjust not used
KYBD Dialing	yes	
Busy Out	no	Yes, if device is accessed through a hunt group
Permit Mismatch	yes	No, if device does not support XON/XOFF flow control

Continued on next page

Table 17. DLDM screen settings for connection to data terminal or personal computer (Continued)

Field on screen	Option	Comments
Parity	Same as DTE	
Dial Echoing	no	These devices can dial in the ASCII stream without human intervention
Disconnect Sequence	Long <BREAK>	-
Answer Text	no	These devices may not want to see any text
Connected Indication	-	Don’t care
Configuration	yes	

Host computers

A host computer may originate and terminate a data call. For this application, the number of DLCs required depends on the number of ports needed. An MADU can be used (instead of 8 ADUs) to complete the connection. Table 18 on page 707 lists option settings for a port that has a terminating connection to a host computer or an originating connection from a host computer.

NOTE:
If the KYBD Dialing field is **n**, the rest of the option settings are irrelevant.

Table 18. DLDM screen settings for terminating connection to host computer

Field on screen	Option	Comments
Speed	All speeds at which the computer can operate	Subject to distance limitations; Autoadjust not used
KYBD Dialing	no	
Busy Out	-	Don’t care
Permit Mismatch	-	Don’t care
Parity	-	Don’t care
Dial Echoing	-	Don’t care
Disconnect Sequence	-	Don’t care
Answer Text	-	Don’t care
Connected Indication	-	Don’t care
Configuration	-	Don’t care

```
change data-module 30                DATA MODULE                      Page 2 of 2
   LAYER 2 PARAMETERS
            Number of Outstanding Frames (w): 4
                  Retry Attempt Counter (N2): 2
                             Frame Size (N1): 135
    Retransmission (T1) Timer (1/10 seconds): 10
              Idle (T4) Timer (1/10 seconds): 30
   LAYER 3 PARAMETERS
     Number of Outstanding Packets: 2
     Restart (T20) Timer (seconds): 8
       Reset (T22) Timer (seconds): 10
```

Screen 76. X.25 Adjunct Data Module

Number of Outstanding Frames (w)

Specifies layer 2 window size.

Valid entries	Usage
1 - 7	If you enter **2**, up to 2 frames can be sent without confirmation.

Retry Attempt Counter (N2)

Specifies the number of times to send one frame when this frame is not confirmed for a period of time.

Valid entries	Usage
0 - 7	

Frame Size (N1)

Specifies the number of bytes in a frame.

Valid entries	Usage
135 **263**	If the value is **135**, there can be up to 1080 bits within a frame. This value is suitable for all adjuncts and for DCS.

Retransmission (T1) Timer (1/10 seconds)

The T1 timer is started at the beginning or the end of the transmission of a frame. At the end of this timer, retransmission of a frame is initiated according to the procedures for link set-up and disconnection or information transfer.

Valid entries	Usage
0 - 250	

Idle (T4) Timer (1/10 seconds)

The T4 timer is a system parameter that represents the time a DTE allows without frames being exchanged on the data link.

Valid entries	Usage
0 - 250	

Number of Outstanding Packets

Specifies the number of packets that can be sent without confirmation.

Valid entries	Usage
2 - 7	

Restart (T20) Timer (seconds)

The T20 timer is a DTE time-limit started when DTE issues a restart indication and terminated when the restart request is received or confirmed.

Valid entries	Usage
0 - 500	

Reset (T22) Timer (seconds)

The T22 timer is a DTE time-limit started when DTE issues a reset indication and terminated when the reset request is received or confirmed.

Valid entries	Usage
0 - 500	

```
change data-module 30                                                    Page 2 of 2
                                 DATA MODULE
  BRI LINK/MAINTENANCE PARAMETERS
                   XID? y      Fixed TEI? n      TEI: ___
         MIM Support? y     Endpt Init? y     SPID: 300_____   MIM Mtce/Mgt? y
```

Screen 77. 7500 and World Class BRI Data Module

XID

(Exchange identification) Used with 7500 and World Class BRI Data Modules. Used to identify layer 2 XID testing capability.

Valid entries	Usage
y/n	Avaya recommends setting to **n**.

Fixed TEI

Used with 7500 and World Class BRI Data Modules. Used to indicate whether the endpoint has Fixed Terminal Equipment Identifier (TEI) capability. TEI identifies a unique access point within a service. For Fixed TEI stations, the TEI must be administered. Terminals with automatic TEI capability, the associated TEI is assigned by the system.

Valid entries	Usage
y/n	Enter **y** to indicate the endpoint has Fixed Terminal Equipment Identifier (TEI) capability.

TEI

Used with 7500 and World Class BRI Data Modules. Appears only if the Fixed TEI field is **y.**

Valid entries	Usage
0 - 63	Enter a 1- to 2-digit number.

MIM Support

Used with 7500 Data Modules. Management Information Message Support. Used to support two types of capabilities: MIM endpoint initialization capability (SPID support), and other Maintenance/Management capability.

Valid entries	Usage
y/n	Enter a 1- to 2-digit number.

Country Protocol

Used with World Class BRI Data Modules. Enter the protocol that corresponds to your supported initialization and codesets. The Country Protocol must match any previously-administered endpoint on the same port. The following table lists the valid protocol entries.

Country/Area	Protocol
Australia	2
ETSI (Europe)	etsi
Japan	3
Singapore	6
United States (Bellcore National ISDN)	1

Endpt Init

Used with 7500 and World Class BRI Data Modules. Endpoint initialization is a procedure, required for multipoint operation, by which User Service Order Profile (USOP) is associated with an endpoint on the ISDN-BRI. This association is made via the Service Profile Identifier (SPID), administered into the system and entered into the ISDN-BRI terminal. For a ISDN-BRI terminal to become operational in a multipoint configuration, both the administered SPID and the SPID programmed into the ISDN-BRI terminal must be the same. This means that the SPID of the new or re-used terminals must be programmed to match the administered SPID value.

Valid entries	Usage
y/n	Indicates the terminal's endpoint initialization capability.

SPID

Used with 7500 and World Class BRI Data Modules. Appears only if the Endpt Init field is **y**. The Service Profile Identifier (SPID) is a variable parameter of up to 10 digits. The SPID must be different for all terminals on the ISDN-BRI and from the Service SPID. The SPID should always be assigned. If the SPID is not assigned for the first ISDN-BRI on a port, any other ISDN-BRI assignment to that port is blocked.

Valid entries	Usage
0 - 9999999999	

MIM Mtce/Mgt

Used with 7500 Data Modules.

Valid entries	Usage
y/n	Entering **y** indicates the terminal supports MIM Maintenance and Management capabilities, other than endpoint initialization.

Endpt ID

Used with World Class BRI Data Modules. Appears only if the Endpt Init field is **y**. This field provides for multipoint configuration conformance to the Bellcore Terminal Initialization procedures. In these procedures, a multipoint configuration requires that the last 2 digits of the Service Profile Identifier (SPID) be between **00** and **63** and be binary unique for each endpoint. This field, combined with the SPID, gives the effective SPID administered into the terminal. Bellcore ISDN-1 requires that the SPID programmed into the endpoint contain at least 9 digits. (For example, if the SPID field is **1234**, and the Endpt ID field is set to **01**, then the SPID administered on the terminal is 000123401. The three leading zeros are necessary to create a 9-digit SPID.)

Valid entries	Usage
00 - 62	Enter a 2-digit number. Each Endpt ID field must have a unique value for each endpoint on the same port.

Date and Time

Use this screen to set the system date and time, to select the daylight savings plan number, if any, and to show whether the current time is standard time or daylight savings. Settings on this screen affect your switch's internal clock and timestamp. You should update the date and time for a leap year or a system restart after a power failure. The correct date and time assure that CDR records are correct. CDR does not work until the date and time have been entered.

For additional information, see the *MultiVantage™ Guide to ACD Call Centers*.

Field descriptions for page 1

```
set time
                           DATE AND TIME
DATE
      Day of the Week: _______     Month: ________
     Day of the Month: __           Year: ____

TIME
     Hour: __   Minute: __    Second: __     Type: ________________

                 Daylight Savings Rule: _

WARNING: Changing the date or time may impact BCMS, CDR, SCHEDULED EVENTS,
         and MEASUREMENTS
```

Screen 78. Date and Time

Day of the Week

Valid entries	Usage
Sunday through **Saturday**	Enter the current day of the week. The system clock uses this as the current day.

Month

Valid entries	Usage
January through **December**	Enter the current month. The system clock uses this as the current month.

Day of the Month

Valid entries	Usage
1 to **31**	Enter the current day of the month. The system clock uses this as the current date.

Year

Valid entries	Usage
1990 to **2099**	Enter the current year. The system clock uses this as the current year.

Hour

The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Valid entries	Usage
0 to **23**	Enter the current hour to be used by the system clock.

Minute

Valid entries	Usage
0 to **59**	Enter the current minute. The system clock uses this as the current minute.

Second

This display-only field shows the seconds and cannot be modified. It resets to zero when you save the information on this screen.

Type

Valid entries	Usage
daylight-savings	Enter **daylight-savings** to indicate daylight savings time is in effect.
standard	Enter **standard** to indicate standard time is in effect.

Daylight Savings Rule

This field displays which daylight savings rule is in use for your system.

Valid entries	Usage
0 to **15**	Enter the appropriate rule number. The system clock uses this as the current daylight savings rule. These rules are defined on the Daylight Savings Rules screen.

Related topics

To update the date and time for the change to or from daylight savings time, use the Daylight Saving Rule screen. Refer to ‘‘Establishing daylight savings rules’’ on page 37 for instructions on how to set up daylight savings rules.

Daylight Savings Rules

Use this screen to enter up to 15 customized daylight savings rules. You can specify the day, month, date, time, and increment each daylight savings rule goes into effect and the day, month, date, and time it stops. Rule 0 makes no adjustment to the system clock for daylight savings and cannot be modified. Telephone displays are affected by these settings.

Field descriptions for page 1

```
change daylight-savings-rules                                Page 1 of 2
DAYLIGHT SAVINGS RULES
Rule           Change Day                Month    Date    Time    Increment

0:   No Daylight Savings

1:   Start: first _________ on or after ________ __ at __:__    ________
     Stop:  first _________ on or after ________ __ at __:__
2:   Start: first _________ on or after ________ __ at __:__
     Stop:  first _________ on or after ________ __ at __:__
3:   Start: first _________ on or after ________ __ at __:__
     Stop:  first _________ on or after ________ __ at __:__
4:   Start: first _________ on or after ________ __ at __:__
     Stop:  first _________ on or after ________ __ at __:__
5:   Start: first _________ on or after ________ __ at __:__
     Stop:  first _________ on or after ________ __ at __:__
6:   Start: first _________ on or after ________ __ at __:__
     Stop:  first _________ on or after ________ __ at __:__
7:   Start: first _________ on or after ________ __ at __:__
     Stop:  first _________ on or after ________ __ at __:__
```

Screen 79. Daylight Savings Rules

Rule

This display-only field indicates the daylight savings rule number.

Change day (Start)

Valid entries	Usage
Sunday through **Saturday or Day**	Enter the day of the week you want the clock to move ahead to begin daylight savings. If you enter **Day** in this field, the clock will change on the exact date entered in the next two fields.

Month (Start)

Valid entries	Usage
January through **December**	Enter the month you want the clock to move ahead to begin daylight savings.

Date (Start)

Valid entries	Usage
0 to **31**	Enter the day of the month you want the clock to move ahead to begin daylight savings.

Time (Start)

The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Valid entries	Usage
0 to **23**	Enter the hour you want the clock to move ahead to begin daylight savings.
0 to **59**	Enter the minute you want the clock to move ahead to begin daylight savings.

Increment (Start)

Valid entries	Usage
0 to **23**	Enter the number of hours you want the clock to move ahead for daylight savings and to move back to return to standard time.
0 to **59**	Enter the number of minutes you want the clock to move ahead for daylight savings and to move back to return to standard time.

Change day (Stop)

Valid entries	Usage
Sunday through **Saturday or Day**	Enter the day of the week you want the clock to move back to return to standard time. If you enter **Day in** this field, the clock will change on the exact date entered in the next two fields.

Month (Stop)

Valid entries	Usage
January through **December**	Enter the month you want the clock to move back to return to standard time.

Date (Stop)

Valid entries	Usage
0 to **31**	Enter the date you want the clock to move back to return to standard time.

Time (Stop)

The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Valid entries	Usage
0 to **23**	Enter the hour you want the clock to move back to return to standard time.
0 to **59**	Enter the minute you want the clock to move back to return to standard time.

Dial Plan Analysis Table

The Dial Plan Analysis Table is the system's guide to translating the digits dialed by users. This screen enables you to determine the beginning digits and total length for each type of call that your switch needs to interpret. The Dial Plan Analysis Table and the Dial Plan Parameters screen work together to define your system's dial plan.

```
change dialplan analysis                                          Page   1 of   3   SPE A
                             DIAL PLAN ANALYSIS TABLE
                                                                     Percent Full:    9


      Dialed  Total  Call       Dialed  Total  Call       Dialed  Total  Call
      String  Length Type       String  Length Type       String  Length Type
        0       1    attd
        1       3    dac
        20      5    ext
        21      2    fac
        3       6    ext

        4       4    ext
        4       7    ext
        5       7    ext
        6       5    ext
        8       1    fac
        9       5    ext
        *       3    fac
        #       3    fac
```

Screen 80. Dial Plan Analysis Table

Percent Full

Displays the percentage (0 to 100) of the system's memory resources that have been allocated for the dial plan that are currently being used.

Dialed String

The dialed string contains the digits that the switch will analyze to determine how to process the call.

Valid entries	Usage
0–9, * and #	Enter up to 2 characters for each call type. * and # can only be the first digit in a string. **NOTE:** For call type attd, if the total length is 2, the Dialed String must be 2 digits long.

Total Length

Valid entries	Usage
1–6 for ars **1–6 for aar** **1–2 for attd** **1–4 for dac** **1–4 for fac** **1–7 for ext** **2–6 for pext**	Enter the number of digits for this call type. The allowed length varies by call type. This must be greater than or equal to the number of digits in the Dialed String.

Call Type

Valid entries	Usage
aar	Automatic Alternate Routing — can be entered only if the Private Networking field on the System-Parameters Customer-Options screen is y. When an aar entry is used to route a call, the caller has access to all AAR features. The Dialed String for **aar** call types must begin with digits 0 – 9. Any extension entry with a shorter length than the **aar** call type is considered during digit analysis. Any extension entry that with a longer total length than the **aar** call type cannot be dialed directly. Instead, an AAR number must be dialed, and digit conversion must be set up to convert the AAR number into an extension.
ars	Automatic Route Selection— can be entered only if the ARS field on the System-Parameters Customer-Option screen is y. When an ars entry is used to route a call, the caller has access to all ARS features. The Dialed String for **ars** call types must begin with digits 0 – 9. You can enter **ars** only once for a given dialed string, and only **ext** or **attd** can begin with the same number. Any extension entry with a shorter length than the **ars** call type is considered during digit analysis. Any extension entry that with a longer total length than the **ars** type call cannot be dialed directly. Instead, an ARS number must be dialed, and digit conversion must be set up to convert the ARS number into an extension.
attd	Attendant — Defines how users call an attendant. Attendant access numbers can start with any number from 0 – 9 and contain 1 or 2 digits. If a telephone's COR restricts the user from originating calls, this user cannot access the attendant using this code.

Valid entries	Usage
dac	Dial access code — Allows you to use trunk access codes (TAC) and feature access codes (FAC) in the same range. Dial access codes can start with any number from 0–9, * or # and can contain up to 4 digits. The system requires that a DAC have the longest total length for a given Dialed String. You can use the DAC to activate or deactivate a switch feature or to seize a trunk from a trunk group, or both. In the first case, the DAC functions as a FAC, in the second as a TAC. For example, you can define the group 300–399 for dial access codes, and allow both FAC and TAC in that range. You can use 4-digit DACs for ordinary trunk access, but they do not work for attendant control of trunk groups, trunk-ID buttons, or DCS, and only the last 3 digits of the codes can be recorded in CDR records. A DAC must be the last item entered in a row when mixed station numbering is used.
ext	Primary extension — Defines extension ranges that can be used on your system. Extension can have a first digit of 0 through 9 and can be 1 – 7 digits in length. Extension cannot have the same first digit as the ARS or AAR feature access code (FAC). It is recommended that you do not administer extensions that begin with **0** if the Uniform Dialing Plan is enabled.

Valid entries	Usage
fac	Feature access code only — A FAC can be any number from 1–9 and contain up to 4 digits. You can use * or #, but only as a first digit. It is recommended that a FAC have the longest total length for a given dialed string when using mixed numbering. Otherwise, problems may occur when 3-digit FACs and 4-digit extensions begin with the same first digit and the FAC is an abbreviated dialing list access code. However, if the entry in the dial plan that defines the FAC is used to define the AAR or ARS access code, then it *must* have the longest total length in the dial plan.
pext	Prefixed extension — Is made up of a prefix (first digit) that can be a **0**–**9** (* and # not allowed) and an extension number of up to 6 digits in length. The maximum length of a prefix and extension combination is 7 digits. You cannot administer a dial access code with the same first digit as a prefixed extension. The purpose of the prefix is to identify the call type as an extension. After digit collection, the prefix digit is removed from the string of dialed digits. The remaining digits (extension number) are then processed. A prefixed extension allows the use of extensions numbers with any dialed string (the extension length must be specified on the table). The “prefixed extension” cannot have the same dialed string as the ARS or AAR facility access code (FAC). When a dial plan has mixed station numbering, extensions of various lengths (all with the same first digit) are mapped on the Dial Plan Analysis table. The system then employs an inter-digit time-out to ensure that all dialed digits are collected. The inter-digit time-out may add several seconds to the dial time. An alternative to the delay required in the time-out mechanism at the expense of dialing an extra digit is to use prefixed extensions in the dial plan.

Dial Plan Parameters screen

The Dial Plan Parameters screen works with the Dial Plan Analysis Table to define your system's dial plan.

```
change dialplan parameters                                          Page 1 of 1
                              DIAL PLAN PARAMETERS

                                                      Local Node Number:  2_
                                                        ETA Node Number:  __
                                                      ETA Routing Pattern:  __
            UDP Extension Search Order: local-extensions-first
```

Screen 81. Dial Plan Parameters screen

Local Node Number

Enter a number to identify a specific node in a switch network. This entry must match the DCS switch node number and the CDR node number if they are specified.

Valid entries	Usage
1–63	Enter the number of a specific node in a network.
blank	The field may be left blank if automatic restoration, DCS, and CDR are not used.

ETA Node Number

Enter the number of the destination switch for Extended Trunk Access (ETA) calls. ETA calls are unrecognized numbers you can send to another switch for analysis and routing. Such numbers can be Facility Access Codes, Trunk Access Codes, or extensions that are not in the UDP table.

Valid entries	Usage
1 – 999	Enter the number of a destination switch.

ETA Routing Pattern

Enter the number of the routing pattern to reach the destination switch.

Valid entries	Usage
1 – 254	Enter the number of the ETA routing pattern

UDP Extension Search Order

Specifies the first table to search to match a dialed extension.

Valid entries	Usage
local-extensions-first	Search the local Dial Plan first to match a dialed extension.
udp-table-first	Search the UDP tables for an off-switch (UDP) conversion.

DCS to QSIG TSC Gateway screen

Use the DCS to QSIG TSC Gateway screen to determine when and how to convert messages from an administered AUDIX NCA-TSC to a QSIG NCA-TSC. This screen maps the AUDIX NCA-TSC to the appropriate machine ID index to find the QSIG subscriber entry in the QSIG MWI-Prefix screen. It also assigns the voice mail number to be used when a DCS served-user node interrogates a QSIG message center.

This screen only appears if the Interworking with DCS field is enabled on the System-Parameters Customer Options screen.

```
change isdn dcs-qsig-tsc-gateway                                  Page 1 of 1

                                DCS TO QSIG TSC GATEWAY

                                  AAR/                                      AAR/
                                  ARS                                       ARS
Mach  Sig   TSC     Voice Mail    Access    Mach  Sig   TSC    VoiceMail   Access
 ID   Grp  Index     Number        Code      ID   Grp  Index    Number      Code
 __   __   ___   ____________     _____     ___   ___  ____   _________     ____
 __   __   ___   ____________     _____     ___   ___  ____   _________     ____
 __   __   ___   ____________     _____     ___   ___  ____   _________     ____
 __   __   ___   ____________     _____     ___   ___  ____   _________     ____
 __   __   ___   ____________     _____     ___   ___  ____   _________     ____
 __   __   ___   ____________     _____     ___   ___  ____   _________     ____
 __   __   ___   ____________     _____     ___   ___  ____   _________     ____
 __   __   ___   ____________     _____     ___   ___  ____   _________     ____
 __   __   ___   ____________     _____     ___   ___  ____   _________     ____
 __   __   ___   ____________     _____     ___   ___  ____   _________     ____
 __   __   ___   ____________     _____     ___   ___  ____   _________     ____
```

Screen 82. DCS to QSIG TSC Gateway screen

Mach ID

You can enter up to 20 machine IDs.

Valid entries	Usage
1 - 20	Enter a unique machine ID. The system does not allow you to specify an ID that you already entered on the Processor Channel screen.

Sig Grp

You must complete the Signaling Group field for each machine ID.

Valid entries	Usage
1 - 110	Enter the assigned signaling group number between 1 and 110 for DEFINITY Server SI
1 -416	Enter the assigned signaling group number between 1 and 416 for DEFINITY Server R

TSC Index

You must complete the TSC Index field for each machine ID.

Valid entries	Usage
1 - 64	Enter the assigned signaling group number for **qsig-mwi** application type on the Signaling Group screen.

Voice Mail Number

This field can be left blank.

Valid entries	Usage
0 - 9	Enter the complete Voice Mail Dial Up number up to 15 digits.

AAR/ARS Access Code

This field can be left blank.

Valid entries	Usage
0 - 9, *, #	Enter up to 4-digit access code.

Digit Absorption

This screen implements up to 5 digit absorption lists. The screen may be required for each CO and FX trunk group connected to a step-by-step CO. Each outgoing digit string from the switch to the step-by-step CO is treated according to entries in the “Absorption Treatment Assignment” section of the screen.

NOTE:
If the Expected Digits field on the Trunk Group screen is blank, you cannot administer Digit Absorption.

Field descriptions for page 1

```
change digit absorption                                            Page 1 of 1
                              DIGIT ABSORPTION

                           List Number: __
  ABSORPTION TREATMENT INFORMATION (All selections must be from same group)
                Choice      Meaning
    Group I.      A       Digit not absorbed.
                  B       Digit absorbed repeatedly.
                  C       Digit absorbed once with no further absorption.
    Group II.     A       Digit not absorbed.
                  D       Digit absorbed only if it is the first digit.
                  E       Digit absorbed only if it is the second digit and
                          the first digit was already absorbed.
                  F       Digit asorbed only if it is the first or second digit.

  ABSORPTION TREATMENT ASSIGNMENT (select treatment (A-F) for each digit below)
          0: A          2: A         4: A         6: A         8: A
          1: A          3: A         5: A         7: A         9: A
```

Screen 83. Digit Absorption

List Number

Displays the Digit Absorption List number (**0** through **4**). The list number is referenced from a field entry on the associated trunk group.

Absorption Treatment Information

This is a display-only section. It shows how Digit Absorption treats each digit, 0 through 9, depending on the assignment of A through C for Group I, and A, D, E, and F for Group II. Enter the assignment on the next section on the screen.

Absorption Treatment Assignment

Enter a desired treatment letter. All choices for the digits 0 through 9 must be taken from the same group (Group I or Group II).

Valid entries	Usage
A through **F**	

DLG Administration

This screen appears if, on the Station screen, the Service Type field is **DLG**. This screen includes all clients that are allowed to connect to the virtual DEFINITY LAN Gateway (DLG).

Field descriptions for page 1

```
                    DLG Administration                   Page X of Y

CTI_-Link    Enabled    Client Name         Client Link    Client Status

    1          y        client1                 1            in use
    3          y        client2                 1            idle
    4          y        client1                 2            idle
```

Screen 84. DS1 Circuit Pack

Enabled

Valid entries	Usage
y/n	Enter **y** to access DLG.

Client Name

A display-only field that indicates the node name of the adjunct requesting service from the DLG.

Valid entries	Usage

Client Link

Valid entries	Usage
	Enter the number between 1 and the number of ASAI links allowed on the system (for instance, 8 or 16).

Client Status

A display-only field indicating **idle** (currently there is no active connection) or **in use** (currently there is an active connection for this client link).

DS1 Circuit Pack

Use this screen to administer all DS1 circuit packs.

Field descriptions for page 1

```
add ds1 xxxxxx                                                   Page 1 of 2
                                DS1 CIRCUIT PACK

             Location: _____                          Name: _______________
             Bit Rate: _____                   Line Coding: ____
    Line Compensation: _                      Framing Mode: ___
       Signaling Mode: ___________               D-Channel: __
              Connect: _________                 Interface: ___________
         Interconnect: ___                   Peer Protocol: ________
    TN-C7 Long Timers? _                  Country Protocol: ____
                                          Protocol Version: _
 Interface Companding: _____                          CRC? _
            Idle Code: ________
                               DCP/Analog Bearer Capability: ________
    MMI Cabling Board: _____  MMI Interface: ESM

MAINTENANCE PARAMETERS

       Slip Detection? _                 Near-end CSU Type: ________
```

Screen 85. DS1 Circuit Pack

```
add ds1 xxxxxx                                                   Page 1 of 2
                                DS1 CIRCUIT PACK

             Location: _____                          Name: _______________
             Bit Rate: _____                   Line Coding: ____
    Line Compensation: _                      Framing Mode: ___
       Signaling Mode: ___________               D-Channel: __
              Connect: _________                 Interface: ___________
         Interconnect: ___                   Peer Protocol: ________
                                          Country Protocol: ____
                                          Protocol Version: _
Interface Companding: _____                           CRC? _
            Idle Code: ________
                               DCP/Analog Bearer Capability: ________
    MMI Cabling Board: _____  MMI Interface: ESM

MAINTENANCE PARAMETERS

       Slip Detection? _                 Near-end CSU Type: ________
```

Screen 86. DS1 Circuit Pack

```
add ds1 xxxxxx                                                   Page 1 of 2
                                DS1 CIRCUIT PACK

             Location: ______                          Name: ____________
             Bit Rate: ______                   Line Coding: ____

       Signaling Mode: __________

         Interconnect: ____                Country Protocol: ______

 Interface Companding: ____
            Idle Code: ____

               Received Digital Metering Pulse Minimum (ms):
               Received Digital Metering Pulse Maximum (ms):
                      Received Digital Metering Pulse Value:
       Slip Detection: __                  Near-end CSU Type: ______
```

Screen 87. DS1 Circuit Pack screen for Croatia and South Africa

The following screen is valid *only* for the TN2242.

```
add ds1 xxxxxx                                                   Page 1 of 2
                                DS1 CIRCUIT PACK

             Location: 01A13                           Name: _______________
             Bit Rate: 2.048                    Line Coding: cmi

       Signaling Mode: CAS
         Interconnect: pbx
                                           Country Protocol: 3

 Interface Companding: mulaw
            Idle Code: 11111111



MAINTENANCE PARAMETERS

       Slip Detection? n
```

Screen 88. DS1 Circuit Pack screen for Channel Associated Signaling

Location

This display-only field shows the port address specified in the **add** command when the circuit pack was first administered.

Name

Use this field to assign a significant, descriptive name to the DS1 link. Avaya recommends putting the vendor’s circuit ID for the link in this field, because that information helps you troubleshoot problems with the link, but you could also use this field to indicate the function or the destination of this DS1 facility. In that case, put the DS1 link circuit ID in the Name field of the trunk group associated with this link.

Valid entries	Usage
1–15 characters	Enter a name for the DS1 link.

Bit Rate

Use this field to select the maximum transmission rate for DS1 circuit packs that support either T-1 or E-1 service. For circuit packs that only support one of these services, the field is a display-only field.

NOTE:
Once an **add ds1** operation is complete (that is, the DS1 screen has been submitted) you can’t change the Bit Rate field with a **change ds1** command. Instead, execute a **remove ds1** command. Then use the **add ds1** command to administer the circuit pack again. You’ll have to re-enter all the information for the circuit pack.

TN464C (and later release) circuit packs have an option switch that must be set to match the entry in the Bit Rate field.

Valid entries	Usage
1.544	Use for T-1 service.
2.048	Use for E-1 service.

Line Coding

This field selects the type of line coding used on this facility. The setting in this field must match the setting on the far-end of the link, or you must have an intervening CSU to convert the line coding protocols. Voice calls will work even if line coding does not match, but a single data call will bring down the DS1 facility. For the TTC 2Mb CMI Trunk circuit pack, this is a display-only field showing **cmi** (coded mark inversion).

The following information is for reference. Talk with your network service provider or your Avaya representative to find the appropriate protocol for your application.

CAUTION:
If you change this field, you must busy out the DS1 circuit pack. You must also change the following screens: Route-Pattern, Access Endpoint, PRI Endpoint, Signaling-Group, and Trunk-Group.

NOTE:
When the DS1 circuit pack is used for ISDN service, the ISDN D-channel data is inverted when **ami-basic** or **ami-zcs** is entered and not inverted when **b8zs** or **hdb3** is entered.

Valid entries	Usage
b8zs (bipolar eight zero substitution)	Enter **b8zs** for T-1 facilities that support voice and/or data traffic. Enter **b8zs** if you need a 64K clear channel.
ami-zcs (alternate mark inversion - zero code suppression)	Enter **ami-zcs** only for T-1 facilities that carry voice traffic: it is not recommended for digital-data applications. If you anticipate upgrading this facility to ISDN, use b8zs line coding if possible.
ami-basic (alternate mark inversion-basic)	Enter **ami-basic** for unrestricted E-1 facilities.
hdb3 (high density bipolar 3)	Enter **hdb3** for restricted E-1 facilities.
cmi (coded mark inversion)	Used in Japan, **cmi** is the only type of line coding you can use with the Japanese 2Mbit trunk circuit pack. This field becomes a display-only field when you are administering the Japanese 2Mbit trunk circuit pack.

Line Compensation

The appropriate entry in this field varies with the type of cable used, so work with your network service provider to determine the correct setting in your situation. The following table shows the appropriate entries for different lengths of 22-gauge ABAM cable terminated on a DSX-1 cross-connect.

Valid entries	Usage
1	Length: 000–133 (ft), 000-40.5 (m)
2	Length: 133–266 (ft), 40.5–81.0 (m)
3	Length: 266–399 (ft), 81.0–122 (m)
4	Length: 399–533 (ft), 122–163 (m)
5	Length: 533–655 (ft), 163–200 (m)

The following table shows the appropriate entries for different lengths of 22-gauge ABAM cable directly connecting to DS1 interfaces.

Valid entries	Usage
1	Length: 0000–0266(ft), 000–081(m)
2	Length: 0266–0532(ft), 081–162(m)
3	Length: 0532–0798(ft), 162–243(m)
4	Length: 0798–1066(ft), 243–325(m)
5	Length: 1066–1310(ft), 325–400(m)

Framing Mode

Use this field to select either superframe (sf or d4) or extended superframe (esf) for T1 service on the DS1 link. The framing mode you use must match the mode used on the other end of the link, so work with your network services provider to determine the appropriate entry for this field.

This field only appears if the Bit Rate field is **1.544** (that is, if you're using T-1 service). If you're using E-1 service, Avaya MultiVantage automatically selects CEPT1 framing.

Tip:

Avaya recommends using ESF when your service provider supports it, especially if you may someday upgrade the facility to ISDN. The ESF format provides enhanced performance measurements and uses a sophisticated error-checking method to ensure data integrity.

Valid entries	Usage
d4	Enter **d4** to use the basic DS1 superframe (sf). This mode is recommended only for voice traffic.
esf	Enter **esf** to use the Extended Superframe format. This mode is recommended for digital data traffic. If you enter **esf** for a TN464F, TN767E, or a later suffix DS1 circuit pack, a second page of the DS1 Circuit Pack screen becomes available to administer ESF Data Link options.

Signaling Mode

This field selects the signaling method used for the DS1 link. This mode must match the method used on the other end of the link, so work with your network services provider to determine the appropriate entry for this field.

Valid entries	Usage
CAS (Channel Associated Signaling)	Enter **CAS** for out-of band signaling with E-1 service. This setting yields 30 64-kbps B-channels for voice or data transmission. Channel 0 is used for framing while channel 16 carries signaling.
robbed-bit	Enter **robbed-bit** for in-band signaling with T-1 service. This setting yields 24 56-kbps B-channels for voice transmission.
isdn-pri	Enter **isdn-pri** for either T-1 or E-1 ISDN service. This setting supports both Facility Associated Signaling and Non-Facility Associated Signaling.
isdn-ext	Enter **isdn-ext** for either T-1 or E-1 ISDN service. This setting supports only Non-Facility Associated Signaling.
common-chan	Enter **common-chan**, for out-of-band signaling with T-1 service. This setting yields 23 64-kbps B-channels for voice or data transmission. Channel 24 is used for signaling.

D-Channel

The Japanese 2Mbit trunk circuit pack, when administered to support ISDN-PRI signaling, allows you to assign the D-channel to any channel from 1 to 31 in an E-1 facility. You cannot submit the screen if this field is blank. Using the **change ds1** command, you can change this field if the D-channel is not used in a signaling group. This field appears only when the Location field indicates the circuit pack is a Japanese 2Mbit trunk circuit pack and the Signaling Mode field is **isdn-pri**.

Valid entries	Usage
1 to **31**	Enter the number of the channel that will be used as the D-channel.

Connect

In order to control communications at layers 2 and 3 of the ISDN-PRI protocol, use this field to specify what is on the far end of this DS1 link. This field only appears when the Signaling Mode field is **isdn-pri**.

Valid entries	Usage
pbx	Enter **pbx** if this DS1 link is connected to another switch in a private network. If **pbx** is entered, the Interface field appears.
line-side	Enter **line-side** when the switch is acting as the network side of an ISDN-PRI interface. Use **line-side** to connect to Roll About Video equipment.
network	Enter **network** when the DS1 link connects this switch to a central office or any other public network switch.
host	Enter **host** when the DS1 link connects this switch to a computer.

Interface

This field only appears when the Connect field is **pbx**; that is, when this DS1 link is providing an ISDN-PRI connection in a private network. The Interface field controls how your switch negotiates glare with the far-end switch. The switches at either end of the DS1 link must have complementary settings in this field: if not, the D-channel won't even come up. For example, if the switch at one end of the link is administered as **network**, the other must be administered as **user**.

Valid entries	Usage
Use the following 2 values for private network applications in the U.S.	
network	Enter **network** if your switch overrides the other end when glare occurs. If you are connecting your switch to a host computer, set this field to **network**.
user	Enter **user** if your switch releases the contested circuit and looks for another when glare occurs. If you are connecting your switch to a public network, set this field to **user**.
Use the following values for private networks (including QSIG networks) outside the U.S. Entering either of these values causes the Peer Protocol and Side fields to appear.	
peer-master	Enter **peer-master** if your switch overrides the other end when glare occurs.
peer-slave	Enter **peer-slave** if your switch releases the contested circuit and looks for another when glare occurs.

Interconnect

For E-1 service using channel-associated signaling, the entry in this field tells Avaya MultiVantage whether the DS1 circuit pack is using a public or private network protocol. The entry in this field must agree with the entry in the Group Type field on the Trunk Group screen. This field appears only when the Signaling Mode field is **CAS**.

Valid entries	Usage
pbx	If **pbx** is selected, the board operates as a tie trunk circuit pack.
CO	If **CO** is selected, the board operates as a CO or DID circuit pack.

TN-C7 Long Timers

This field controls the T302 and T303 timers. It only appears if the Signaling Mode field is **isdn-pri**.

Valid entries	Usage
y	Use if you want to increase the length of the long timers.
n	Leave n if you want to use the default long timers.

Country Protocol

The entry in this field must match the country protocol used by the far-end switch. For connections to a public network, your network service provider can tell you which country protocol they are using.

This field appears if the Signaling Mode field is **CAS** or **isdn-pri**. For the Japanese 2Mbit trunk circuit pack, this is a display-only field if the Signaling Mode field is **CAS**.

Valid entries	Usage
1 to **25**	Enter the country protocol used by the central office at which this link terminates.
etsi	Enter **etsi** if your network service provider uses the protocol of the European Telecommunications Standards Institute (ETSI). Enter **etsi** only if the Signaling Mode field is **isdn-pri**.

Protocol Version

In countries whose public networks allow multiple layer-3 signaling protocols for ISDN-PRI service, this field selects the protocol that matches your network service provider's protocol. Refer to ''Public network signaling administration for ISDN-PRI Layer 3'' on page 737 to see which countries support which protocols.

This field appears only when:

- The Signaling Mode field is **isdn-pri** and the Connect field is **network.**
- The Signaling Mode field is **isdn-pri,** the Connect field is **pbx**, and the Interface field is **user** or **network.**

Valid entries	Usage
a, b, c, d	The entry in this field must match the protocol used by your network service provider, so work with your vendor to determine the appropriate entry.

WARNING:

The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/protocol-version combination on the DS1 screen, you cannot set the Send Connected Number field to **r** *(restricted) on the ISDN-PRI Trunk Group screen, as this causes display problems.*

Public network signaling administration for ISDN-PRI Layer 3

The table below shows Avaya MultiVantage public network access connections for ISDN-PRI Layer 3.

Admin value	Country	Protocol supported	B-channel mtce msg
1-a	United States, Canada	AT&T TR 41449/ 41459 (tested with AT&T network, Canadian network, and MCI network)	Service
1-b	United States	Bellcore TR 1268; NIUF.302; ANSI T1.607	Restart
1-c	United States	NORTEL DMS-250 BCS36/IEC01	Service
1-d	United States	Telecordia SR-4287	Service
2-a	Australia	AUSTEL TS014.1; Telecom Australia TPH 1856 National ISDN protocol	Restart
2-b	Australia	ETSI ISDN protocol	Restart
3	Japan	NTT INS-NET	Restart
4	Italy	ETS 300 102	Restart
5	Netherlands	ETS 300 102	Restart
6	Singapore	ETS 300 102	Restart
7	Mexico	ETS 300 102	Restart
8	Belgium	ETS 300 102	Restart
9	Saudi Arabia	ETS 300 102	Restart
10-a	United Kingdom	ETS 300 102 (for connection to DASS II/DPNSS through external converter)	Restart

Continued on next page

Admin value	Country	Protocol supported	B-channel mtce msg
10-b	United Kingdom, Ireland	ETS 300 102 (Mercury); British Telecom ISDN 30; Telecom Eireann SWD 109	none
11	Spain	Telefonica ISDN Specification	Restart
12-a	France	VN4 (French National PRI)	None
12-b	France	ETS 300 102 modified according to P10-20, called Euronumeris	None
13-a	Germany	FTZ 1 TR 6 (German National PRI)	None
13-b	Germany	ETS 300 102	Restart
14	Czech Republic, Slovakia	ETS 300 102	Restart
15	Russia (CIS)	ETS 300 102	Restart
16	Argentina	ETS 300 102	Restart
17	Greece	ETS 300 102	Restart
18	China	ETS 300 102	Restart
19	Hong Kong	ETS 300 102	Restart
20	Thailand	ETS 300 102	Restart
21	Macedonia	ETS 300 102	Restart
22	Poland	ETS 300 102	Restart
23	Brazil	ETS 300 102	Restart
24	Nordic	ETS 300 102	Restart
25	South Africa	ETS 300 102	Restart
ETSI-a	Europe, New Zealand, etc.	ETS 300 102	Restart
ETSI-b		ETS 300 102	None

Peer Protocol

This allows you to administer the peer level protocol that will operate in a private network. This field appears if the Interface field is **peer-master** or **peer-slave**. To enter **Q-SIG**, the Basic Call Setup field on the System-Parameters Customer-Options screen must be **y**.

Valid entries	Usage
Q-SIG	This implements QSIG Network Basic Call.
TTC	For private networking. Requires a Digital Trunk (Japan 2 MB TTC) (TN2242) circuit pack.

Interworking Message

This field determines what message the switch sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group).

Valid entries	Usage
PROGress	Normally select this value. PROGress asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk.
ALERTing	ALERTing causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

Side

This field controls how your switch resolves glare at layer 3 over an ISDN-PRI link in QSIG private networks. It appears if the Interface field is **peer-master** or **peer-slave**.

The default value of the field changes depending upon which value the Interface field contains.

CAUTION:

It is critical that administration on this switch correctly pairs with the administration of the far-end switch. If the far-end is administered as the b side, this field should be set to a regardless of whether the layer 2 designation is peer-master or peer-slave, and vice versa.

Valid entries	Usage
a	Enter **a** if the Interface field is **peer-master** (this switch overrides the far-end when glare occurs).
b	Enter **b** if the Interface field is **peer-slave** (this switch releases the contested circuit and looks for another when glare occurs).

Protocol Version

In countries whose public networks allow multiple layer 3 signaling protocols for ISDN-PRI service, this field selects the protocol that matches your network service provider's protocol. Refer to ''Public network signaling administration for ISDN-PRI Layer 3'' on page 1545 to see which countries support which protocols.

This field appears only when:

- The Signaling Mode field is **isdn-pri** and the Connect field is **network.**
- The Signaling Mode field is **isdn-pri,** the Connect field is **pbx**, and the Interface field is **user** or **network.**

Valid entries	Usage
a **b** c	The entry in this field must match the protocol used by your network service provider, so work with your vendor to determine the appropriate entry.

WARNING:

The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/protocol-version combination on the DS1 screen, you cannot set the Send Connected Number field to ***r*** *(restricted) on the ISDN-PRI Trunk Group screen, as this causes display problems.*

Interface Companding

The entry in this field must match the companding method used by the far-end switch. This field does not appear for all DS1 circuit packs.

Valid entries	Usage
alaw	Enter **alaw** for E-1 service.
mulaw	Enter **mulaw** for T-1 service.

CRC

This field indicates whether a cyclic redundancy check (CRC) will be performed on transmissions that the DS1 circuit pack receives. This field does not display for all circuit packs.

Valid entries	Usage
y	Enter **y** when the Signaling Mode field is **CAS** and the DS1 link is providing E-1 service.
n	Enter **n** for all other applications.

Idle Code

CAUTION:

Customers: The entry in the Country Protocol field sets the default idle code. Do not change the default without assistance from Avaya or your network services provider.

For some circuit packs, this is a display-only field.

Valid entries	Usage
any 8-digit string of 0's and 1's	This entry sets the signal sent out over idle DS0 channels. The string must be compatible with the protocol used by the far-end switch.

Channel Numbering

The ETSI and ISO QSIG specifications require that B-channels on an E1 be encoded as 1-30 in the Channel ID IE. Prior to this field, Avaya MultiVantage only used this scheme for Country Protocols 2a (Australia) and 13a (Germany 1TR6). This field appears when the Signaling Mode field is **isdn-pri**, the Bit Rate field is **2.048**, the Connect field is **pbx**, and the Interface field is **peer-master** or **peer-slave**.

Valid entries	Usage
timeslot	
sequential	If the Avaya MultiVantage is connected via QSIG trunks to a switch supporting the ETSI QSIG or ISO QSIG specifications, this field must be **sequential**. When the Signaling Mode field is **isdn-pri** and the Bit Rate field is **2.048**, but the Channel Numbering field does not display because of the setting of other fields, it is set internally to **sequential** for 2a (Australia) and 13a (Germany).

DMI-BOS

The DMI/BOS protocol is used for high-speed digital communications between a host computer and an Avaya MultiVantage. With this 24-channel protocol, channels 1–23 of the DS1 link carry data and channel 24 carries control signaling. DMI/BOS has greater capacity than a robbed-bit 24-channel facility. This field appears only when the Signaling Mode field is **common-chan**.

Valid entries	Usage
y	Enter **y** to activate the Digital Multiplexed Interface-Bit Oriented Signaling (DMI-BOS) format.
n	Enter **n** to use an Avaya proprietary format.

DCP/ANALOG Bearer Capability

This field appears when the Signaling Mode field is **isdn-pri**. It is used to determine bearer capability encoding.

Valid entries	Usage
3.1kHz	
speech	

MMI Cabling Board

This field appears only if the MMCH field is **y** on the System-Parameters Customer-Options screen.

Valid entries	Usage
slot address (cabinet, carrier, slot)	Enter the slot location (cabinet, carrier, slot) of the multimedia interface circuit pack that is connected to the Expansion Services Module (ESM).

MMI Interface

This display-only field appears if the MMCH field is **y** on the System-Parameters Customer-Options screen and there is a value in the MMI Cabling Board field.

Received Digital Metering Pulse Minimum (ms)

This field appears only when the Signal Mode field is **cas** (Channel Associated Signaling), the Interconnect field is **co** or **pbx**, and the Country Protocol field is administered for a protocol that uses periodic pulse metering (PPM) as defined in Table 19 on page 744. The default value depends on the Country Protocol field's entry.

Valid entries	Usage
20 to **1000** ms in increments of 10ms.	Work with your network services provider to determine the appropriate entry. The entry must be less than the Received Digital Metering Pulse Maximum field.

Received Digital Metering Pulse Maximum (ms)

This field appears only when the Signal Mode field is **cas** (Channel Associated Signaling), the Interconnect field is **co** or **pbx**, and the Country Protocol field is administered for a protocol that uses periodic pulse metering (PPM) as defined in Table 19. The default value depends on the Country Protocol field's entry.

Valid entries	Usage
20 to **1000** ms in increments of 10ms.	Work with your network services provider to determine the appropriate entry. The entry must be greater than the Received Digital Metering Pulse Minimum field.

Received Digital Metering Pulse Value

This field appears when the Signal Mode field is **cas** (Channel Associated Signaling), the Country Protocol field is **21**, and the Interconnect field is **co** or **pbx**.

Valid entries	Usage
0, **1**	Work with your network services provider to determine the appropriate entry.

Table 19. Incoming Digital PPM Signaling Default (per Country Protocol code)

Code	Country	PPM Min (ms)	PPM Max (ms)	PPM Value
0	null	NA	NA	NA
1	U.S.	NA	NA	NA
2	Australia	80	180	0
3	Japan	NA	NA	NA
4	Italy	120	150	1
5	Netherlands	90	160	0
6	Singapore	NA	NA	NA
7	Mexico	20	180	1
8	Belgium	20	180	1
9	Saudi Arabia	NA	NA	NA
10	UK	NA	NA	NA
11	Spain	20	220	0
12	France	NA	NA	NA
13	Germany	NA	NA	NA
14	Czech Republic	20	420	1
15	Russia CIS	NA	NA	NA
16	Argentina	10	180	1
17	Greece	100	180	1
18	China	NA	NA	NA
19	Hong Kong	NA	NA	NA

Continued on next page

Table 19. Incoming Digital PPM Signaling Default (per Country Protocol code) (Continued)

Code	Country	PPM Min (ms)	PPM Max (ms)	PPM Value
20	Thailand	20	180	1
21	Macedonia	120	180	1
	Croatia	20	80	1
22	Poland	100	150	0
23	Brazil	NA	NA	NA
24	Nordic	NA	NA	NA
25	South Africa	160	240	0, 1

Slip Detection

Slips — synchronization errors — slow digital transmissions and can result in data loss. The switch maintains a slip-count record for each DS1 interface to detect errors and evaluate their severity (the type of alarm). If as many as 50 percent of those spans administered for slip detection are experiencing slips (with respect to the primary), then a decision is made to switch to the secondary.

CAUTION:

Always enter **y** *for DS1 circuit packs that serve as primary or secondary synchronization references.*

Valid entries	Usage
y	Enter **y** to allow maintenance software to measure the slip-rate of this circuit pack and determine whether it's excessive. Typically, enter **y** for DS1 spans used for data applications and for spans used as synchronization references. This excludes all T1-spans connecting channel banks, unless the channel bank is externally timed. This entry enables switching between the primary and secondary synchronization references and an internal high-accuracy clock.
n	Enter **n** for DMI-BOS links or when testing is not required. Normally, enter **n** for DS1 spans that are used exclusively for voice and that do not serve as the primary or secondary synchronization source.

Near-end CSU Type

This field appears only when the DS1 circuit pack is a TN767D or TN464E or later suffix model, the Bit Rate field is **1.544** and the Country Protocol field is **1** (U.S.). This field does not display for all circuit packs.

Valid entries	Usage
other	Enter **other** if no channel service unit is attached to the DS1 facility or if the CSU is an external unit. No options are available on page 2 for administering an external CSU.
integrated	Enter **integrated** if a 120A CSU module is attached to the DS1 board. This integrated channel service unit (ICSU) can accept software-administrable option downlinks (that is, it can respond to test codes from technician's equipment and report its status). When you enter **integrated**, fields for administering options on the ICSU appear on page 2 of the DS1 Circuit Pack screen.

Alarm When PRI Endpoint Detached

Use this field for DS1 circuit packs connected to Roll-About Video equipment. This field appears only when the Connect field is **line-side**.

Valid entries	Usage
y/n	Enter **y** if you want the switch to generate an alarm when the DS1 board detects a loss of signal (for example, if the video equipment is disconnected).

Echo Cancellation

Appears when DS1 Echo Cancellation is **y** on the System-Parameters Customer-Options screen and circuit packs support echo cancellation.

Valid entries	Usage
y/n	Enter **y** to allow echo cancellation.

EC Direction

Direction of echo cancellation. Appears when Echo Cancellation is **y** on the DS1 Circuit Pack screen.

Valid entries	Usage
inward/outward	Indicate the direction of the echo that is being cancelled.

EC Configuration

Appears when Echo Cancellation is **y** on the DS1 Circuit Pack screen.

Valid entries	Usage
1-15	Four default echo cancellation configurations are available: Enter 1, 5-15: provides the most rapid adaptation in detecting and correcting echo at the beginning of a call. Enter 2: slightly slower adaptation to echo, use if speech is clipped when both parties talk at the same time. Enter 3: slightly slower adaptation to echo, may result in a 2 or 3 second fade on strong echo Enter 4: in cases of extreme echo, excessive clipping or breakup of speech. May result in slight echo or background noise.

Field descriptions for page 2

```
add ds1 xxxxxx
                               DS1 CIRCUIT PACK

ESF DATA LINK OPTIONS

                      Network Management Protocol:
  Send ANSI-T1.403 One-Second Performance Reports?
                              Far-end CSU Address:


INTEGRATED CSU OPTIONS
                                     Transmit LBO:
                                     Receive ALBO:
                                     Upon DTE LOS:

CPE LOOPBACK JACK OPTIONS
                   Supply CPE Loopback Jack Power?
```

Screen 89. DS1 Circuit Pack

CAUTION:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Page 2 does not appear for all DS1 circuit packs. For those circuit packs that support it, this page appears only when the Framing Mode field is **esf** or the Near-end CSU Type field is **integrated**.

Network Management Protocol

This field appears only if the Framing Mode field is **esf**.

Valid entries	Usage
tabs	The entry in this field, used only with circuit packs that have an integrated channel service unit (CSU), allows the data link to be remotely monitored.

Send ANSI-T1.403 One-Second Performance Reports

This field selects whether your DS1 circuit pack will send error reports to the far-end switch. These reports are useful for network management, and are sent at 1-second intervals when enabled. This field appears only if the Framing Mode field is **esf**. It is used only with circuit packs that have an integrated channel service unit (CSU).

Valid entries	Usage
y/n	Enter **n**. Consult your Avaya representative if you think you may want to use these reports.

Far-end CSU Address

This field, which, appears only if the Framing Mode field is **esf**.

Valid entries	Usage
a **b**	Enter **b**. This field administers the transmit direction address used for the **ESF data link** command with both integrated and external channel service units (CSU).

Transmit LBO (Transmit Line Build-Out)

This field reduces the outgoing signal strength by a fixed amount. The appropriate level of loss depends on the distance between your switch (measured by cable length from the smart jack) and the nearest repeater. Where another switch is at the end of the circuit, as in campus environments, use the cable length between the 2 switches to select the appropriate setting from the table below. This field appears if the Near-end CSU Type field is **integrated**.

Valid entries	Usage
0db	For distances of 2,001–3,000 feet
-7.5db	For distances of 1,001–2,000
-15db	For distances of 0–1,000 feet
-22.5db	For mid-span repeaters

Receive ALBO (Receive Automatic Line Build-Out)

This field increases the strength of incoming signals by a fixed amount to compensate for line losses.

Valid entries	Usage
26db **36db**	To set this field correctly, you should measure the signal loss on this specific facility. However, you may enter **26db** for most applications. **36db** is occasionally appropriate, mainly on campus networks that don’t conform to public telephone network standards.

Upon DTE LOS

DTE stands for “Data Terminal Equipment.” This field tells the switch what to do if the outgoing signal from the DS1 circuit pack (the data terminal equipment) to the network is lost.

Valid entries	Usage
loopback	Enter **loopback** to return the network signal to the network. This prevents any alarms at the far-end switch.
ais	Enter **ais** (Alarm Indicator Signal) to send an unframed all-ones signal (the AIS or Blue Alarm) to the far-end switch. This option alerts your network service provider to the problem immediately and aids troubleshooting.

Supply CPE Loopback Jack Power

If a CPE (Customer Premise Equipment) Loopback Jack is installed, the DS1 board should supply power to the equipment during loopback testing.

Valid entries	Usage
y/n	Enter **y** if a CPE Loopback Jack is installed. If not, you must enter **n**.

Related topics

Refer to ‘‘Setting up digital trunks’’ on page 417 for instructions.

Refer to ‘‘DS1 Trunk Service’’ on page 1542 for detailed information.

Extended Pickup Group

This screen allows grouping of pickup groups into extended pickup groups. This allows users to answer calls outside their immediate group. The maximum number of groups that can be added to an extended pickup group is 25.

Field descriptions

```
change extended-pickup-group 1                                     Page   1 of   2
                              EXTENDED PICKUP GROUP

                            Extended Group Number: 56

Pickup   Pickup Group                           Pickup   Pickup Group
Number     Number                               Number     Number

 0:      _____                                  13:      _____
 1:      _____                                  14:      _____
 2:      _____                                  15:      _____
 3:      _____                                  16:      _____
 4:      _____                                  17:      _____
 5:      _____                                  18:      _____
 6:      _____                                  19:      _____
 7:      _____                                  20:      _____
 8:      _____                                  21:      _____
 9:      _____                                  22:      _____
10:      _____                                  23:      _____
11:      _____                                  24:      _____
12:      _____                                  25:      _____
13:
```

Screen 90. Extended Pickup Group screen

Extended Group Number

This display-only field shows the number associated with a collection of pickup groups. The extended group is a collection of pickup groups that can answer calls from other pickup groups in the same extended group.

Pickup Number

This display-only field shows the pickup number assigned to the pickup group number. This is the number users dial after the feature access code (FAC) to pick up calls in their extended pickup group.

Pickup Group Number

This field determines which call pickup groups can answer calls in the extended pickup group.

Valid entries	Usage
1-800 (DEFINITY Server SI) **1-5000** (DEFINITY Server R)	Enter the pickup group numbers for each of the pickup groups that you want to belong to this extended group.

Extensions Administered to have an MCT-Control Button

This screen lists the extensions that can take control of a Malicious Call Trace (MCT) request. In order to give a user the ability to take control of such requests, you need to add their extension to this list and assign them a mct-control feature button.

Field descriptions for page 1

```
  change mct-group-extensions
                                                                Page 1 of 1
Extensions Administered to have an MCT-Control Button:
  1: ________ 19: ________ 37: ________ 55: ________ 73: ________  91: ________
  2: ________ 20: ________ 38: ________ 56: ________ 74: ________  92: ________
  3: ________ 21: ________ 39: ________ 57: ________ 75: ________  93: ________
  4: ________ 22: ________ 40: ________ 58: ________ 76: ________  94: ________
  5: ________ 23: ________ 41: ________ 59: ________ 77: ________  95: ________
  6: ________ 24: ________ 42: ________ 60: ________ 78: ________  96: ________
  7: ________ 25: ________ 43: ________ 61: ________ 79: ________  97: ________
  8: ________ 26: ________ 44: ________ 62: ________ 80: ________  98: ________
  9: ________ 27: ________ 45: ________ 63: ________ 81: ________  99: ________
 10: ________ 28: ________ 46: ________ 64: ________ 82: ________ 100: ________
 11: ________ 29: ________ 47: ________ 65: ________ 83: ________
 12: ________ 30: ________ 48: ________ 66: ________ 84: ________
 13: ________ 31: ________ 49: ________ 67: ________ 85: ________
 14: ________ 32: ________ 50: ________ 68: ________ 86: ________
 15: ________ 33: ________ 51: ________ 69: ________ 87: ________
 16: ________ 34: ________ 52: ________ 70: ________ 88: ________
 17: ________ 35: ________ 53: ________ 71: ________ 89: ________
 18: ________ 36: ________ 54: ________ 72: ________ 90: ________
```

Screen 91. Malicious Call Trace control extensions

1–100

Enter the extension for a telephone or attendant console that you want to have an MCT-Control button. Note that you must also assign the `mct-control` button on the extension's Station or Attendant Console screen.

Feature Access Code

This screen assigns feature access codes (FACs) that, when dialed, activate or cancel the system features. Each field on this screen has the same valid values, which must conform to feature access codes or dial access codes as defined by your dial plan.

Valid entries	Usage
1–4 digit number, *** #**	***** and **#** may be used as first digit. However, analog rotary dial phones cannot use the “*” and “#” symbols.

Field descriptions for page 1

```
change feature-access-codes                                       Page 1 of 7
                                   FEATURE ACCESS CODE (FAC)
             Abbreviated Dialing List1 Access Code: ____
             Abbreviated Dialing List2 Access Code: ____
             Abbreviated Dialing List3 Access Code: ____
    Abbreviated Dial - Prgm Group List Access Code: ____
                          Announcement Access Code: ____
                           Answer Back Access Code: ____
          Auto Alternate Routing (AAR) Access Code: ____
           Auto Route Selection (ARS) Access Code1: ____   Access Code 2: ____
                     Automatic Callback Activation: ____    Deactivation: ____
    Call Forwarding Activation Busy/DA: ____    All: ____    Deactivation: ____
                             Call Park Access Code: ____
                           Call Pickup Access Code: ____
    CAS Remote Hold/Answer Hold-Unhold Access Code: ____
                      CDR Account Code Access Code: ____
                            Change COR Access Code: ____
                       Change Coverage Access Code: ____
                      Data Origination Access Code: ____
                          Data Privacy Access Code: ____
                    Directed Call Pickup Access Code: ____
```

Screen 92. Feature Access Code (FAC) screen

Abbreviated Dialing List1 Access Code

Used to access AD list 1.

Abbreviated Dialing List2 Access Code

Used to access AD list 2.

Abbreviated Dialing List3 Access Code

Used to access AD list 3.

Abbreviated Dial - Prgm Group List Access Code

Used to enter a group list from a telephone. The user’s extension must be entered on the Abbreviated Dial Group List screen in order to program the group list.

Announcement Access Code

Used to record announcements.

Answer Back Access Code

Used to retrieve parked calls.

Auto Alternate Routing (AAR) Access Code

Used to access AAR.

Auto Route Selection (ARS) Access Code1

Used to access ARS. You can have one ARS access code for local and one for long distance, and route accordingly.

(ARS) Access Code 2

Also used to access ARS.

Automatic Callback Activation/Deactivation

Used to activate/cancel Automatic Callback.

Call Forwarding Activation Busy/DA

Used to forward calls to an administered number if the user is busy or does not answer.

(Call Forwarding Activation) All

Used to forward calls to an administered number for all calls.

(Call Forwarding) Deactivation

Used to deactivate call forwarding.

Call Park Access Code

Used to park an active call, which can then be retrieved from a different station using the answer back access code. Do not administer to have the same first digit as another feature access code that is longer in length.

Call Pickup Access Code

Used to answer a call directed to a pickup group.

CAS Remote Hold/Answer Hold-Unhold Access Code

Used by a Centralized Attendant Service (CAS) attendant to place calls on hold and answer calls held at a remote switch.

CDR Account Code Access Code

Used prior to entering an account code for CDR purposes.

Change COR Access Code

Used to allow users to change their class of restriction (COR) from a phone. This field can only be used if the Change COR by FAC field is enabled on the System-Parameters Customer-Options screen.

Change Coverage Access Code

Used to change a coverage path from a telephone or remote station.

Data Origination Access Code

Used to originate a data call from a voice station.

Data Privacy Access Code

Used to isolate a data call from call waiting or other interruptions.

Directed Call Pickup Access Code

Used to establish directed call pickup.

Field descriptions for page 2

```
change feature-access-codes                                        Page  2  of 7
                            FEATURE ACCESS CODE (FAC)
     Emergency Access to Attendant Access Code:
                     Enhanced EC500 Activation:        Deactivation:
     Extended Call Fwd Activate Busy D/A_  All:        Deactivation:
        Extended Group Call Pickup Access Code:
               Facility Test Calls Access Code:
                             Flash Access Code:
              Group Control Restrict Activation:        Deactivation:
                     Hunt Group Busy Activation:        Deactivation:
                              ISDN Access Code:
                Last Number Dialed Access Code:
    Leave Word Calling Message Retrieval Lock:
   Leave Word Calling Message Retrieval Unlock:
             Leave Word Calling Send A Message: #66
           Leave Word Calling Cancel A Message: *66
               Malicious Call Trace Activation:        Deactivation:
         Meet-me Conference Access Code Change:
PASTE  (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:        Dissociate Code:
        Per Call CPN Blocking Code Access Code:
      Per Call CPN Unblocking Code Access Code:
```

Screen 93. Feature Access Code (FAC) screen

Emergency Access To Attendant Access Code

Used to gain access to the attendant in an emergency situation. Such calls alert as emergency calls. This field cannot be used if the Emergency Access to Attendant field is not enabled on the System-Parameters Customer-Options screen.

Enhanced EC500 Activation

Used to activate extension to cellular where calls to an office number are extended to a cell phone.

Extended Call Fwd Activate Busy D/A

Used to activate call forwarding from a telephone or remote location.

Extended Call Fwd Activate All

Used to activate call forwarding from a telephone or remote location.

Extended Call Fwd Deactivation

Used to deactivate call forwarding from a telephone or remote location.

NOTE:
An extension must have Station Security Codes administered to use the following FACs:

- Extended Call Forward All Activate
- Extended Call Forward Busy/Don’t Answer Activate
- Extended Call Forward Deactivate
- Change Coverage

Extended Group Call Pickup Access Code

The feature access code (FAC) users enter when a call directed to another pickup group is to be answered. Users must enter a valid “Pickup Number” following the Extended Group Call Pickup Access Code to complete the operation.

Facility Test Calls Access Code

Used to place activate a facility test call.

SECURITY ALERT:
To ensure the security of your system, leave Facility Test Calls Access Code blank except when actually testing trunks.

Flash Access Code

Used to generate trunk flash. This code ensures that the flash signal is interpreted by the central office switch, rather than the Avaya MultiVantage.

Group Control Restrict Activation / Deactivation

Used to change the restriction level for all users with a given class of restriction. Requires console permissions.

Hunt Group Busy Activation/Deactivation

Hunt Group members can dial these codes to place themselves in a busy state, and to become available again.

ISDN Access Code

Used to place an ISDN call without using ARS, AAR, or UDP.

Last Number Dialed Access Code

Used to redial the last number dialed from this station.

Leave Word Calling Message Retrieval Lock

Used to lock the display module on telephones. The lock function activates at a telephone by dialing this system-wide lock access code. This prevents unauthorized users from displaying, canceling, or deleting messages associated with the telephone. The Lock Messages field on the Station screen also must be enabled.

Leave Word Calling Message Retrieval Unlock

Used to unlock a telephone’s display module. The lock function is canceled at the telephone by dialing this unlock FAC followed by the SCC.

Leave Word Calling Send A Message

Used to send a leave word calling message.

Leave Word Calling Cancel A Message

Used to cancel a leave word calling message.

Malicious Call Trace Activation

Used to activate a trace request on a malicious call.

Meet-me Conference Access Code Change

Allows the controlling user of a Meet-me Conference VDN to change the access code.

PASTE (Display PBX data on Phone) Access Code

Allows users to view call center data on display phones. PASTE is used in conjunction with Avaya IP Agent.

Personal Station Access (PSA) Associate Code

Used to associate a telephone with the phone features assigned to a user’s extension.

Personal Station Access (PSA) Dissociate Code

Used to remove the association between a physical phone and an extension number. You cannot provide the code until Personal Station Access (PSA) on the System Parameters Customer-Options screen is **y**.

Per Call CPN Blocking Code Access Code

If CPN blocking is off for a trunk group, users can turn it on for a call by using this code. When they dial this code, the calling party number is not sent to the public network.

Per Call CPN Unblocking Code Access Code

If CPN blocking is on for a trunk group, users can turn it off for a call by using this code. When they dial this code, the calling party number is sent to the public network.

Field descriptions for page 3

```
                                                                   Page 3 of 7
                                   FEATURE ACCESS CODE (FAC)

                  Priority Calling Access Code:
                           Program Access Code:
       Refresh Terminal Parameters Access Code:
              Remote Send All Calls Activation:       Deactivation:
              Self Station Display Access Code:
                     Send All Calls Activation:       Deactivation:
         Station Firmware Download Access Code:
                       Station Lock Activation:       Deactivation:
      Station Security Code Change Access Code:
                          Telephone Activation: #*
             Terminal Dial-up Test Access Code:
Terminal Translation Initialization Merge Code:       Separation Code:
            Transfer to Voice Mail Access Code:
          Trunk Answer Any Station Access Code:
              User Control Restrict Activation:       Deactivation:
  Voice Coverage Message Retrieval Access Code:
 Voice Principal Message Retrieval Access Code: *80
```

Screen 94. Feature Access Code (FAC) screen

Program Access Code

Used to program abbreviated dial buttons on an individual phone.

Refresh Terminal Parameters Access Code

Used to update terminal parameters on an individual phone when system settings have changed.

Remote Send All Calls Activation/Deactivation

Used to activate or deactivate the Send All Calls feature. Requires console permissions.

Self Station Display Activation

The self station field is not active. If set to a valid FAC, a digital station displays its primary extension number when the FAC is entered.

Send All Calls Activation/Deactivation

Used to activate or deactivate sending all calls to coverage with minimal or no alerting at the station.

Station Firmware Download Access Code

This field specifies the feature access code used for 2420 DCP station firmware downloads.

Station Lock Activation/Deactivation

Used to activate or deactivate Station Lock.

Station Security Code Change Access Code

Enter the code the user must dial to change their Station Security Code. The SCC must be administered before the user can change it using this FAC. That is, a user cannot change a blank SCC.

Terminal Dial-Up Test Access Code

Used to perform tests on digital telephones to make sure that the telephone and the buttons are communicating properly with the switch. See your Maintenance documentation for information about Digital Terminal Remote Looparound Test.

Terminal Translation Initialization Merge Code

Enter the digits that must be dialed to install (merge) a station without losing any of its previous feature settings. The Terminal Translation Initialization Separation Code must have been used, or an X administered in the Port field of the Station screen, when the telephone was removed from its former location in order for the Terminal Translation Initialization Merge Code to be effective. (If you try to use this and it doesn’t work, check the station screen for this extension. If there is still a port assigned, type X in the port field, then try the TTI merge again.)

Terminal Translation Initialization Separation Code

Enter the digits that must be dialed to remove (separate) a station from a location without losing any of its feature settings.

Transfer to Voice Mail Access Code

Enter the digits that must be dialed to allow coverage to transfer the caller to the original call recipient's AUDIX mail where the caller can leave a message. Do not administer this code to have the same first digit as another feature access code that is longer in length.

Trunk Answer Any Station Access Code

Enter the access code that station users must dial to answer calls alerting on night bells.

User Control Restrict Activation/Deactivation

Used to change the restriction level for a specific extension. Requires console permissions.

Voice Coverage Message Retrieval Access Code

Allows users to retrieve voice messages for another user (for whom they are a coverage point) via a digital display module.

Voice Principal Message Retrieval Access Code

Allows users to retrieve their own voice messages for another user via a digital display module.

Field descriptions for page 4

```
                                                          Page 4 of 7
                             FEATURE ACCESS CODE (FAC)
          Whisper Page Activation Access Code:
```

Screen 95. Feature Access Code (FAC) screen

Whisper Page Activation Access Code

Allows users to place a page to another user's phone, when active on a call. The paged user, and not the other parties on the call, hears the page.

Field descriptions for page 5

The feature access codes on this page pertain only to ACD call centers.

```
change feature-access-codes                                        Page 5 of 7
                        FEATURE ACCESS CODE (FAC)

                  Automatic Call Distribution Features

                   After Call Work Access Code: ___
                            Assist Access Code: ___
                           Auto-In Access Code: ___
                          Aux Work Access Code: ___
                             Login Access Code: ___
                            Logout Access Code: ___
                         Manual-In Access Code: ___
     Service Observing Listen Only Access Code: ___
     Service Observing Listen/Talk Access Code: ___
                   Add Agent Skill Access Code: ___
                Remove Agent Skill Access Code: ___
            Remote Logout of Agent Access Code: ___

                 Call Vectoring/Call Prompting Features

                     Converse Data Return Code: ___
```

Screen 96. Feature Access Code (FAC) screen

After Call Work Access Code

Enter the code the agent must dial when the agent will be performing work-related ACD activities.

Assist Access Code

Enter the digit the agent must dial to request assistance from the split supervisor.

Auto-In Access Code

Enter the code the agent must dial to become automatically available to receive another ACD call each time a call is released.

Aux Work Access Code

Enter the code the agent must dial when the agent will be performing non-ACD activities.

Login Access Code

Enter the code the agent must dial to gain access to the ACD functions. This is a system-wide code for all ACD agents.

Logout Access Code

Enter the logout code the agent must enter to exit ACD. This is a system-wide logout code for all ACD agents.

Manual-In Access Code

Enter the code the agent must dial to receive a single, new ACD call upon the completion of an ACD call.

NOTE:
The following two fields appear only if Service Observing (Remote/By FAC) on the System Parameters Customer-Options screen is **y**.

Service Observing Listen Only Access Code

Enter the code that must be dialed to allow a station with Service Observing permission (COR) to listen to other agent ACD calls without being heard on the ACD call.

Service Observing Listen/Talk Access Code

Enter the code that must be dialed to allow a station with Service Observing permission (COR) to both listen and be heard on an ACD call.

NOTE:
The following two fields appear only if Expert Agent Selection (EAS) Enabled is optioned on the Feature-Related System-Parameters screen.

Add Agent Skill Access Code

Enter the digits an agent must dial to be able to add a skill to their current skill set.

Remove Agent Skill Access Code

Enter the digits an agent must dial to be able to remove a skill from their current skill set.

NOTE:
The next field is available only if Vectoring (Basic) and Vectoring (Prompting) have been enabled on the System-Parameters Customer-Options screen.

Remote Logout of Agent Access Code

Enter the digits you need to dial to remotely logout an idle ACD or EAS agent.

Converse Data Return Code

Enter the access code the CONVERSANT must outpulse prior to outpulsing the digits being returned to the system. This FAC must match the code administered on CONVERSANT.

Field descriptions for page 6

The feature access codes on this page pertain only to Hospitality features.

```
change feature-access-codes                                    Page 6 of 7
                              FEATURE ACCESS CODE (FAC)

                                Hospitality Features

                          Automatic Wakeup Call Access Code: *11___
                Housekeeping Status (Client Room) Access Code: ______
                Housekeeping Status (Client Room) Access Code: ______
                Housekeeping Status (Client Room) Access Code: ______
                Housekeeping Status (Client Room) Access Code: ______
                Housekeeping Status (Client Room) Access Code: ______
                Housekeeping Status (Client Room) Access Code: ______
                    Housekeeping Status (Station) Access Code: ______
                    Housekeeping Status (Station) Access Code: ______
                    Housekeeping Status (Station) Access Code: ______
                    Housekeeping Status (Station) Access Code: ______
                       Verify Wakeup Announcement Access Code: ______
                             Voice Do Not Disturb Access Code: ______
```

Screen 97. Feature Access Code (FAC) screen

Automatic Wakeup Call Access Code

Enter the access code the user must dial to schedule or cancel a wakeup call.

Housekeeping Status (Client Room) Access Code

Enter the access code the housekeeper dials from the client's room to provide room status. These codes are transmitted to the Property Management System (PMS) for processing. You can assign a definition to the six codes on the Hospitality screen.

Housekeeping Status (Station) Access Code

Enter the access code the housekeeper must dial to provide room status. This access code must be dialed from designated telephones. There are four codes.

Verify Wakeup Announcement Access Code

Enter the access code the user can dial to verify a wakeup announcement.

Voice Do Not Disturb Access Code

Enter the access code the user must dial to enter or cancel a do not disturb request without using a display — through the use of voice prompting.

Field descriptions for page 7

The feature access codes on this page pertain only to Multimedia Call Handling (MMCH).

```
change feature-access-codes                                           Page 7 of 7
                              FEATURE ACCESS CODE (FAC)
                                 Multimedia Features

                      Basic Mode Activation:
                   Enhanced Mode Activation:
                Multimedia Call Access Code:
      Multimedia Data Conference Activation:  Deactivation:
       Multimedia Multi-Address Access Code:
           Multimedia Parameter Access Code:
```

Screen 98. Feature Access Code (FAC) screen

Basic Mode Activation

If you enter this FAC when your system is an Enhanced multimedia complex, it will revert to a Basic multimedia complex. If you enter this FAC when your system is a Basic mode station it will do nothing.

Enhanced Mode Activation

If you enter this FAC when your system is a Basic multimedia complex, it will become an Enhanced multimedia complex. If you enter this FAC when your system is an Enhanced mode station it will do nothing.

Multimedia Call Access Code

If you enter this FAC from any voice station, it indicates to the Avaya MultiVantage that you are making an Enhanced mode multimedia call. If you originate a multimedia call with the multimedia call access code, it will originate a call according to the Default Multimedia Parameters selected on the Feature Related System Parameters screen.

Multimedia Data Conference Activation

If you enter this FAC from any voice station that is participating in a multimedia call, it will alert the Avaya MultiVantage that you want to enable data collaboration with the other parties on the call. If you enter this FAC a second time, it will give denial treatment (since a collaboration session is already active). This FAC only applies to voice stations on an Avaya MultiVantage switch equipped with an ESM adjunct.

Multimedia Data Conference Deactivation

If you enter this FAC from the phone that enabled data collaboration on a multimedia mode call, it will deactivate the data session and revert to a voice and video call. If a user enters this FAC while participating in a data-collaboration multimedia call that the user did not initiate, the system responds with denial treatment.

Multimedia Multi-Address Access Code

The multimedia multi-address access code is similar to the multimedia call access code. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. For example, ISDN-BRI provided by a Central Office is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered.

Originating a multimedia call with the multimedia multi-address access code will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen.

Multimedia Parameter Access Code

This FAC can be entered by any voice station to indicate to the Avaya MultiVantage that you want to initiate a multimedia mode call with a specific bearer capability. This FAC would be followed by a 1 or 2 to indicate the following parameter selections respectively: 2x64 (unrestricted initial system default), 2x56 (restricted).

Feature-Related System Parameters

This screen implements system parameters associated with various System features.

NOTE:
This screen used to contain Call Coverage and Call Forwarding parameters. These fields have been moved to a new screen, which you can access with the command **change system-parameters coverage-forwarding**.

Field descriptions for page 1

```
change system-parameters features  (page 1)
                         FEATURE-RELATED SYSTEM PARAMETERS
                            Self Station Display Enabled? n
                                 Trunk-to-Trunk Transfer? none
 Automatic Callback - No Answer Timeout Interval (rings): 4_
                   Call Park Timeout Interval (minutes): 10
     Off-Premises Tone Detect Timeout Interval (seconds): 20_
                               AAR/ARS Dial Tone Required? y
           Music/Tone On Hold: music               Port: _______
           Music (or Silence) On Transferred Trunk Calls: all
                        DID/Tie/ISDN Intercept Treatment: attd

 Internal Auto-Answer of Attd-Extended/Transferred Calls? y
               Automatic Circuit Assurance (ACA) Enabled? y
                                      ACA Referral Calls: local
                                ACA Referral Destination: ________
            ACA Short Holding Time Originating Extension: ______
             ACA Long Holding Time Originating Extension: ______
           Abbreviated Dial Programming by Assigned Lists:
     Auto Abbreviated/Delayed Transition Interval(rings):
                 Protocol for Caller ID Analog Terminals: Bellcore
 Display Calling Number for Room to Room Caller ID Calls?
```

Screen 99. Feature-Related System Parameters screen

Self Station Display Enabled

Use this field to control the use of the INSPECT button for digital display phones.

Self Station Display allows a user to display the primary extension associated with a digital display phone. There are two methods: (1) enter a feature access code (FAC), and (2) use the INSPECT button. In either case, the display shows the primary extension associated with the phone where the FAC or NORMAL or EXIT button is entered. In the case of the FAC, the display continues until a display-altering event occurs (for instance, going on-hook or receiving an incoming call). In the case of the INSPECT button, the display continues until the user presses the NORMAL or EXIT button or until a display-altering event occurs.

Valid entries	Usage
y	The primary extension does display when the INSPECT button is pressed.
n	The extension does not display when the INSPECT button is pressed.

Trunk-to-Trunk Transfer

Regulations in some countries control the settings for this field. See your Avaya representative for assistance.

Valid entries	Usage
all	Enter **all** to enable all trunk-to-trunk transfers. This allows telephone users to set up trunk-to-trunk transfer, go on-hook without disconnecting the call, and forward the call to a remote location.
restricted	Enter **restricted** (restricted public) to restrict all public trunks (CO, WATS, FX, CPE, DID, and DIOD).
none	Enter **none** to restrict all trunks (except CAS and DCS) from being transferred.

Automatic Callback — No Answer Timeout Interval (rings)

Valid entries	Usage
2–9.	Enter the number of times the callback call rings at the calling station before the callback call is canceled.

Call Park Timeout Interval (minutes)

Valid entries	Usage
1–90.	Enter the number of minutes a call remains parked before it cancels.

Off-Premises Tone Detect Timeout Interval (seconds)

Valid entries	Usage
5–25	The number of seconds a call progress tone receiver (CPTR) tries to detect dial tone from a trunk during dialing. Once the time-out interval occurs, the call either outpulses on the trunk or gets intercept treatment depending on the setting of the Outpulse Without Tone field on page 6 of this screen.

AAR/ARS Dial Tone Required

A second dial tone provides feedback to the user that additional dialing can occur.

Valid entries	Usage
y/n	Enter **y** to indicate a second dial tone is to be given to the calling party on a incoming tie or DID trunk call that is to be routed via AAR/ARS.

Music/Tone on Hold

If you use equipment that rebroadcasts music or other copyrighted materials, you may be required to obtain a copyright license from, or pay fees to, a third party such as the American Society of Composers, Artists, and Producers (ASCAP) or Broadcast Music Incorporated (BMI). You can purchase a Magic OnHold® system, which does not require such a license, from Avaya. This field does not appear if Tenant Partitioning is **y** on the System-Parameters Customer-Options screen. In that case, use the Tenant screen to establish Music on Hold.

Valid entries	Usage
music **tone** **none**	Indicates what a caller hears while on hold.

Port

Indicates the port number that provides Music-on-Hold access. This requires a port on a TN763 Auxiliary Trunk circuit pack or any supported Analog Line circuit pack. Appears when Music/Tone on Hold is **music**.

Valid entries	Usage
01 through **22**	First and second characters are cabinet number
01 through **03**	
A through **E**	Third character is carrier
01 through **20**	Fourth and fifth characters are slot number
01 through **04** (Aux Trunk Port)	Sixth and seventh characters are circuit number
01 through **16** (Analog Port)	

Music (or Silence) On Transferred Trunk Calls

Valid entries	Usage
all	Enter **all** to allow all transferred trunk calls to receive music until the call is answered if the Music-on-Hold feature is available.
no	Enter **no** if trunk callers are to hear music (or silence if Music-on-Hold is not administered) while waiting to be transferred, and then ringback as soon as the transfer is completed till the call is answered.
call-wait	Enter **call-wait** if trunk calls transferred to stations that require the call to wait hear music (if administered); all other transferred trunk calls receive ringback tone.

DID/Tie/ISDN Intercept Treatment

Valid entries	Usage
A recorded announcement extension	Toll charges do not apply to DID and private network calls routed to an announcement.
attd	For system security, Avaya recommends entering **attd** in this field. This routes intercept calls to the attendant and, if the attendant receives several of these, they will know a problem exists.

Internal Auto-Answer of Attd-Extended/Transferred Calls

This only applies to digital telephones (except BRI) with a headset or speakerphone capability.

Valid entries	Usage
attd-extended	Enter **attd-extended** to enable IAA for only attendant extended calls.
both	Enter **both** to enable IAA for station transferred and attendant extended calls.
none	Enter **none** to disable IAA for all calls.
transferred	Enter **transferred** to enable IAA for only station transferred calls.

Automatic Circuit Assurance (ACA) Enabled

If Automatic Circuit Assurance (ACA) Enabled is **n**, associated ACA fields will not display.

Must have an “aca-halt” button administered on the user’s station. If you enable this feature, complete the following ACA-related fields.

Valid entries	Usage
y	Enter **y** if ACA measurements will be taken.
n	Otherwise, enter **n**.

ACA Referral Calls

Indicates where ACA referral calls generate. (Only appears when ACA Enabled is **y**.)

Valid entries	Usage
local	Local referral calls generate on and for the local switch.
primary	Primary referral calls generate on the local switch for remote switches as well as the local switch.
remote	Remote referral calls generate at another switch in a DCS network. In this case, the remote node number must also be entered. The remote node number is the same node number as defined on the Dial Plan screen. Also, ACA button status transmits to other switches when in a DCS network.

ACA Referral Destination

The specified extension should be equipped with a display module. This field only appears if ACA Referral Calls is **local** or **primary**.

Valid entries	Usage
An extension	Enter the extension on the local switch that is to receive the ACA referral call.
attd	Enter **attd** for attendant.

ACA Short Holding Time Originating Extension and ACA Long Holding Time Originating Extension

Valid entries	Usage
An unassigned extension	Do not use the same extension number for both fields. The specified extensions are assigned automatically by the system when the screen is submitted. These fields only display if ACA Referral Calls is **local** or **primary**.

ACA Remote PBX Identification

This field only appears if ACA Referral Calls is **remote**.

Valid entries	Usage
1–63	Enter a number to identify the switch in a DCS network that makes the referral call. Do not define the remote switch identified in this field as **local** on the system's Dial Plan screen.

Abbreviated Dial Programming by Assigned Lists

Valid entries	Usage
y	Enter **y** to allow programming by station's assigned list.
n	Enter **n** if using Program Access code to indicate which personal list is to be programmed.

Auto Abbreviated/Delayed Transition Interval (rings)

Valid entries	Usage
1–16	Enter the number of rings before an automatic abbreviated/ delayed transition is triggered for a call

Protocol for Caller ID Analog Terminals

Determines the protocol/tones sent to a Caller ID phone.

Valid entries	Usage
Bellcore	Enter **Bellcore** for Bellcore protocol with 212 modem protocol tones. Used in the U.S. and similar countries.
V23-Bell	Enter **V23-Bell** for Bellcore protocol with V.23 modem tones. Used in Bahrain and similar countries.

Display Calling Number for Room to Room Caller ID Calls

Valid entries	Usage
y/n	Enter **y** to display the calling number for room to room hospitality calls.

Field descriptions for page 2

```
change system-parameters features  (page 2)
                       FEATURE-RELATED SYSTEM PARAMETERS
LEAVE WORD CALLING PARAMETERS
  Maximum Number of External Calls Logged Per Station: 0
  Message Waiting Indication for External Calls? n
  Stations with System-wide Retrieval Permission (enter extension)
    1: 34430___ 3: attd____ 5: ________ 7: ________ 9: ________
    2: 34412___ 4: ________ 6: ________ 8: ________ 10: ________
WARNING!   SEE USER DOCUMENTATION BEFORE CHANGING TTI STATE
                     Terminal Translation Initialization (TTI) Enabled? _
                    TTI State:              TTI Security Code: _______
                  Customer Telephone Activation (CTA) Enabled?
               Record CTA/PSA/TTI Transactions in History Log? _
                                 COR for PSA Dissociated Sets: _
                            CPN, ANI for PSA Dissociated Sets: _
               Prohibit Bridging Onto Calls with Data Privacy? _
                    Enhanced Abbreviated Dial Length (3 or 4)? _
                Record All Submission Failures in History Log? _
                    Record PMS/AD Transactions in History Log?_
                       Record IP Registrations in History Log?
        Default Multimedia Outgoing Trunk Parameter Selection: 2x64
```

Screen 100. Feature-Related System Parameters screen

Maximum Number of External Calls Logged Per Station

When an external call is not answered, the switch keeps a record of up to 15 calls (provided information on the caller identification is available) and the phone's message lamp lights. The phone set displays the names and numbers of unsuccessful callers.

Valid entries	Usage
0 - 15	The maximum number of calls that can be logged for each user. The assigned number cannot be larger than the entry in the Maximum Number of Messages Per Station (when MSA not in service) field.

Message Waiting Indication for External Calls

Provides a message waiting indication when external calls are logged.

Valid entries	Usage
y	The message waiting indication for a particular station is on whenever an external call is logged.
n	The log of external calls has no impact on the message waiting indication.

Stations With System-wide Retrieval Permission (enter extension)

An extension must be removed from this list before the station is removed from the system. The switch refers to the extensions on this list as “super-retrievers.”

Valid entries	Usage
An assigned extension.	Enter up to 10 telephone extension numbers that can retrieve LWC Messages or External Call Log records for all other telephones. A VDN extension is not allowed.
attd	An entry of **attd** gives retrieval permission to all attendants.

Terminal Translation Initialization (TTI) Enabled

Terminal Translation Initialization (TTI) must be enabled on the System Parameters Customer Options screen before the TTI and Automatic Customer Telephone Rearrangement (ACTR) fields can be administered.

Valid entries	Usage
y	Enter **y** to start ACTR, TTI, and PSA transactions (extension and phone moves between ports).
n	Enter **n** to remove existing TTI port translations and make sure no new TTI port translations are generated.

TTI State

The value of this field determines what type of TTI default port translation is generated for unadministered digital ports. This field appears when Terminal Translation Initialization (TTI) Enabled is **y**.

Valid entries	Usage
data	Used for moving processor data modules, should be turned off after move.
resume	Allows the TTI state to return to what it was before TTI was manually suspended.
suspend	Allows any current generation or removal of TTI port translations to be halted and access to TTI is denied.
voice	Enter voice to allow Personal Station Access and Automatic Customer Telephone Rearrangement.

TTI Security Code

(This is also known as the TTI authorization code but is not the same as system authorization codes.) This field appears when Terminal Translation Initialization (TTI) Enabled is **y**.

Valid entries	Usage
1- to 7-digit code	This is the code that TTI users must enter when accessing TTI from their telephones

Customer Telephone Activation (CTA) Enabled

Valid entries	Usage
y/n	If **y**, you can use #* to automatically activate telephones after administration. See Installing new phones for more information.

Record CTA/PSA/TTI Transactions in History Log

Use this field to record when extensions and physical phones move between ports without additional administration from the switch administrator.

Valid entries	Usage
y/n	Enter **y** to start records for ACTR, TTI, and PSA transactions (extension and phone moves between ports) in the history log. You access these transactions via the List History command.

COR for PSA Dissociated Sets

Valid entries	Usage
1-95	Enter the class of restriction to apply to calls made from dissociated telephones. See ''Personal Station Access'' on page 1674 for more information.

CPN, ANI for PSA Dissociated Sets

Appears when the COR for PSA Dissociated Sets field is non-blank. Specifies the ISDN calling party number (CPN), R2-MFC ANI, and CAMA CESID applied to calls made from PSA dissociated sets, if no system-wide calling party information has been administered for those protocols on their respective administration screens.

Valid entries	Usage
1-20 digits	Enter the calling party number or automatic number identification for calls made from dissociated telephones.

Prohibit Bridging Onto Calls with Data Privacy

Valid entries	Usage
y/n	Enter **y** to protect calls from bridge-on by any party, including Service Observing, Intrusion, Verify, and Bridging.

Enhanced Abbreviated Dial Length (3 or 4)

The administrator may not be able to use all entry slots because of system capacity constraints.

Valid entries	Usage
3	A value of **3** makes 1000 Enhanced List entries available to the administrator
4	A value of **4** makes 10,000 entries available.

Record All Submission Failures in History Log

Allows submission failures to be recorded on the history log.

Valid entries	Usage
y/n	Enter **y** to record submission failures on the history log.

Record PMS/AD Transactions in History Log

Allows PMS and abbreviated dialing button transactions to be recorded on the history log.

Valid entries	Usage
y/n	Enter **y** to record PMS or abbreviated dialing button transactions on the history log.

Record IP Registrations in History Log

Allows the logging of IP registrations in the history log.

Valid entries	Usage
y/n	Enter **y** to record IP registrations on the history log.

Default Multimedia Outgoing Trunk Parameter Selection

Valid entries	Usage
2x56	Sets default parameter for bandwidth and bearer for all video calls.
2x64	

Field descriptions for page 3

```
change system-parameters features  (page 3)
                      FEATURE-RELATED SYSTEM PARAMETERS
Reserved Slots for Attendant Priority Queue: 5_
                                        Time Before Off-Hook Alert: 10__
                            Emergency Access Redirection Extension: ________
              Number of Emergency Calls Allowed in Attendant Queue: __
                                              Call Pickup Alerting? n
                       Temporary Bridged Appearance on Call Pickup? y
                                     Call Pickup on Intercom Calls? y
                                              Directed Call Pickup? n
                                        Extended Group Call Pickup: flexible
                 Deluxe Paging and Call Park Timeout to Originator? n
     Controlled Outward Restriction Intercept Treatment: tone________  ________
    Controlled Termination Restriction (Do Not Disturb): tone________
              Controlled Station to Station Restriction: tone________
AUTHORIZATION CODE PARAMETERS           Authorization Code Enabled? y
                                         Authorization Code Length: 7
                            Authorization Code Cancellation Symbol? #
                                           Attendant Time Out Flag? n
                                        Display Authorization Code? _
                     Controlled Toll Restriction Replaces: station-station
          Controlled Toll Restriction Intercept Treatment: extension     3000
```

Screen 101. Feature-Related System Parameters screen

Reserved Slots for Attendant Priority Queue

Valid entries	Usage
2–**75**	Enter the number of calls that can go in to the emergency queue

Time Before Off-Hook Alert

Valid entries	Usage
1 to **3000** seconds	Enter the time in seconds that a telephone with an Off-Hook Alert Class of Service can remain off-hook (after intercept tone has started) before an emergency call is sent to the attendant.

Emergency Access Redirection Extension

Valid entries	Usage
An assigned extension	Enter the assigned extension number (can be a VDN) where emergency queue overflow will redirect.

Number of Emergency Calls Allowed in Attendant Queue

Valid entries	Usage
0–**75**	Enter the number of calls allowed in the attendant queue before additional calls are routed to the backup extension.

Call Pickup Alerting

This provides pickup group members with a visual indication on the Call Pickup status lamp of calls eligible to be answered via Call Pickup.

Valid entries	Usage
y/n	Enter **y** to enable Call Pickup Alerting on a system-wide basis.

Temporary Bridged Appearance on Call Pickup

Valid entries	Usage
y	Enter **y** to allow a temporary bridged appearance for calls answered with the Call Pickup or Directed Call Pickup features. This field controls this capability on a system-wide basis.
n	Enter **n** to prevent the temporary bridged appearance of calls answered with these features.

Call Pickup on Intercom Calls

Valid entries	Usage
y	Enter **y** to allow a user's or Agent LoginID's call, ringing as an intercom call, to be picked up using the Call Pickup or Directed Call Pickup features. This field controls the use of this feature throughout the system.
n	Enter **n** to prevent the use of these features to pickup an intercom call.

Directed Call Pickup

Feature use by individual stations, attendants, or EAS agents can be controlled by COR.

Valid entries	Usage
y	Enter **y** to allow use of the Directed Call Pickup feature across the system.
n	Enter **n** to prevent feature use.

Extended Group Call Pickup

Enables call pickup groups to answer calls directed to another call pickup group.

Valid entries	Usage
flexible	Flexible feature version supporting a one-to-n (pickup group-to-extended pickup group) mapping.
simple	Simple feature version with a one-to-one pickup group-to-extended pickup group mapping supported.
none	Extended group call pickup not supported.

Deluxe Paging and Call Park Timeout to Originator

Paged calls that are to be parked require separate activation of the Call Park feature. All parked calls that time out return to the attendant.

Valid entries	Usage
y	Enter **y** to enable the Loudspeaker Paging **-** Deluxe feature that essentially integrates the Loudspeaker Paging and Call Park features. All parked calls that time out (not answered by paged party) return to the parking party.
n	Enter **n** to enable the Loudspeaker Paging feature.

Controlled Outward Restriction Intercept Treatment

Enter the type of intercept treatment the caller receives when the call is outward restricted.

Valid entries	Usage
announcement	Provides a recorded announcement to calls that cannot be completed as dialed. You select and record the message. The calling party receives indication that the call is receiving Intercept Treatment. Enter the extension number for the announcement in the associated field.
attendant	Allows attendants to provide information and assistance to outgoing calls that cannot be completed as dialed or that are transferred to incomplete or restricted stations.
extension	Enter the extension number for the extension in an associated field. May not be a VDN extension.
tone	Provides a siren-type tone to internal calls that cannot be completed as dialed

Controlled Termination Restriction (Do Not Disturb)

Enter the type of intercept treatment the caller receives when the call is placed to a termination restricted telephone.

Valid entries	Usage
announcement	If **announcement** is entered, complete an associated extension number field.
attendant	Redirects intercepted calls to the attendant.
coverage	Redirects intercepted calls to coverage.
extension	If **extension** is entered, complete an associated extension number field. May not be a VDN extension,
tone	Provides a siren-type tone to calls that cannot be completed as dialed.

Controlled Station-to-Station Restriction

Enter the type of intercept treatment the caller receives when the call is placed to a restricted telephone.

Valid entries	Usage
announcement	If announcement is entered, an associated extension number field displays. Enter the extension of the restricted telephone in the field.
attendant	Intercepted calls are redirected to the attendant.
extension (may not be a VDN extension)	If extension is entered, an associated extension number field displays. Enter the extension of the restricted telephone in the field.
tone	Intercepted calls receive intercept (siren) tone.

Authorization Code Parameters

Authorization Codes Enabled

This field cannot be administered if Authorization Codes is not enabled on the System-Parameters Customer-Options screen.

SECURITY ALERT:

To maintain system security, Avaya recommends that Authorization Codes be used.

Valid entries	Usage
y/n	Enter **y** to enable Authorization Codes on a systemwide basis.

Authorization Code Length

This field only appears and must be completed if Authorization Codes Enabled is **y**. This is the number of digits that must be assigned to the Authorization Code (AC) field on the Authorization Code screen.

SECURITY ALERT:

You enhance your system's security by using the maximum length for your authorization code.

Valid entries	Usage
4–13 digits	Enter a number that defines the number of digits (length) in the Authorization Code field.

Authorization Code Cancellation Symbol

Enter the symbol a caller must dial to cancel the 10-second wait period during which the user can enter an authorization code. This field only appears when Authorization Code is **y**.

Valid entries	Usage
#	Enter the cancellation code **#** if the main and tandem switches are both the same type of switch.
1	Enter the cancellation code **1** if an Avaya System 85 or DIMENSION PBX switch is part of the complex/network.

Attendant Time Out Flag

If this field is not enabled, the caller receives Intercept tone. This flag affects only remote users or incoming calls over trunks requiring an authorization code. This field only appears if Authorization Codes Enabled is **y**.

Valid entries	Usage
y/n	Enter **y** if a call is to be routed to the attendant if the caller does not dial an authorization code within 10 seconds or dials an invalid authorization code.

Display Authorization Code

This field applies only to DCP, not to BRI or hybrid sets.

SECURITY ALERT:

*To enhance your system's security, set Display Authorization Code to **n**.*

Valid entries	Usage
y	Enter **y** to allow authorization code digits to display on the set during the dialing.
n	Enter **n** if these digits should not display.

Controlled Toll Restriction Replaces

This field activates the Controlled Toll Restriction feature.

Valid entries	Usage
outward **station-station** **none**	The value that you choose for this field will be replaced by controlled toll restriction. In other words, if you choose station-station, you will not be able to use station-station restrictions unless you reset this field.

Controlled Toll Restriction Intercept Treatment

This field applies an intercept treatment to a toll call during the call processing.

Valid entries	Usage
announcement	A sub-field appears to the right if **announcement** is used. If the entry is **announcement**, enter the assigned announcement extension.
attendant	Intercepted calls are redirected to the attendant.
extension	A sub-field appears to the right if **extension** is used. If the entry is **extension**, enter the extension assigned to station or individual attendant.
tone	Intercepted calls receive intercept (siren) tone.

Field descriptions for page 4

```
change system-parameters features  (page 4)
                          FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
              System Printer Endpoint: ____                Lines Per Page: 60
                  EIA Device Bit Rate: 9600

SYSTEM-WIDE PARAMETERS
                          Switch Name: ________________
         Emergency Numbers - Internal:          External: 911
      No-License Incoming Call Number:

MALICIOUS CALL TRACE PARAMETERS
               Apply MCT Warning Tone? n     MCT Voice Recorder Trunk Group: ___
      Delay Sending RELease (seconds)?
SEND ALL CALLS OPTIONS
           Send All Calls Applies to: station
      Auto Inspect on Send All Calls? n

UNIVERSAL CALL ID
     Create Universal Call ID (UCID)? n
                                             UCID Network Node ID: ___
```

Screen 102. Feature-Related System Parameters screen

System Printer Parameters

The system printer is the printer dedicated to support scheduled reports.

System Printer Endpoint

NOTE:
The **eia** option is not available for DEFINITY Server R.

Valid entries	Usage
Data module extension	Associated with the System printer
eia	If the DCE jack is used to interface the printer.
SYS_PRNT	Use this value if the system printer is connected over a TCP/IP link, and the link is defined as SYS_PRNT on the IP Services screen.

Lines Per Page

Valid entries	Usage
24 – **132**	Enter the number of lines per page required for the report.

EIA Device Bit Rate

This field is not displayed for DEFINITY Server R.

Valid entries	Usage
1200 **2400** **4800** **9600**	Enter the required printer speed setting.

System-Wide Parameters:

Switch Name

Valid entries	Usage
Any keyboard character.	Enter alpha-numeric characters for identification.

Emergency Numbers

Enter the phone numbers you want to use for emergency calls. If your system is in a No-License mode, these will be the only numbers that users can dial. The number may contain the feature access code for Emergency Access to the Attendant, trunk access codes, or any number, *, or #.

No-License Incoming Call Number

Enter the administered extension that can receive incoming calls when the switch is in No-License mode.

Malicious Call Trace Parameters:

Apply MCT Warning Tone

Valid entries	Usage
y/n	Enter **y** to provide an audible tone to the controlling station when an MCT recorder is actively recording a malicious call.

MCT Voice Recorder Trunk Group

Assign the trunk group for MCT voice recorders.

Valid entries	Usage
1 to **99**	group number

Delay Sending RELease (seconds)

This field specifies the amount of time DEFINITY waits before sending an ISDN release message in response to receiving an ISDN disconnect message. This field appears only if, on the System-Parameters Customer-Options screen, the Malicious Call Trace field is **y**.

Valid entries	Usage
0-30	Enter the number in increments of 10.

Send All Calls Options:

Send All Calls Applies to

Valid entries	Usage
station	If set to **station**, any call to that station, regardless of the number dialed, causes calls to that station's own extension to be sent immediately to Coverage, or causes calls to different extensions assigned to the station as bridged appearances to become Ring-Ping notification if Redirect Notification field is **y**.
extension	When set to **extension**, only the calls sent to that extension are placed to coverage.

Auto Inspect on Send All Calls

Valid entries	Usage
y	If set to **y**, allows you to be presented automatically with Calling Party information for calls which are silently alerting their station because of the Send-All-Calls feature.
n	If set to **n**, you are not guaranteed a Calling Party display for calls sent directly to Coverage by the Send-All-Calls feature.

Universal Call ID (UCID):

Create Universal Call ID (UCID)

Valid entries	Usage
y	If set to **y**, DEFINITY will generate UCID for each call when necessary.
n	If set to **n**, the DEFINITY will not generate a UCID for any call.

UCID Network Node ID

Enter a number unique to the switch in a network of switches.

Valid entries	Usage
1 to **32767**	This number is an important part of the UCID tag and must be unique to the switch.

Field descriptions for page 5

```
change system-parameters features  (page 5)
                          FEATURE-RELATED SYSTEM PARAMETERS
       Public Network Trunks on Conference Call: 5           Auto Start? n
    Conference Parties with Public Network Trunks: 6            Auto Hold? n
 Conference Parties without Public Network Trunks: 6       Attendant Tone? y
         Night Service Disconnect Timer (seconds): 180      Bridging Tone? n
                 Short Interdigit Timer (seconds): 3      Conference Tone? n
              Unanswered DID Call Timer (seconds): ____    Intrusion Tone? n
              Line Intercept Tone Timer (seconds): 30   Special Dial Tone? n
                 Long Hold Recall Timer (seconds):     Mode Code Interface? n
                      Reset Shift Timer (seconds): 0
     Station Call Transfer Recall Timer (seconds): 0
                               DID Busy Treatment: tone
        Invalid Number Dialed Intercept Treatment: Announcement _____
               Allow AAR/ARS Access from DID/DIOD? _
                 Allow ANI Restriction on AAR/ARS? _

                  7405ND Numeric Terminal Display? n          7434ND? n
DISTINCTIVE AUDIBLE ALERTING
              Internal: 1   External: 2   Priority: 3
                              Attendant Originated Calls:_ ________
    DTMF Tone Feedback Signal to VRU - Connection: _    Disconnection: _
```

Screen 103. Feature-Related System Parameters screen

Public Network Trunks on Conference Call

Indicates the number of public network trunks allowed on a conference call.

Valid entries	Usage
0 to **5**	If this field is **0**, the Conference Parties with Public Network Trunks field will not appear on the screen.

Auto Start

If this field is enabled, the START buttons on all attendant consoles are disabled.

Valid entries	Usage
y/n	Enter **y** to enable the Automatic Start feature.

Conference Parties with Public Network Trunks

Specifies the maximum number of parties allowed in a conference call involving a public network subscriber. If the value of the Public Network Trunks on Conference Call field is **0**, this field will not appear on the screen.

Valid entries	Usage
3 to **6**	

Auto Hold

Valid entries	Usage
y/n	Enter **y** to enable the Automatic Hold feature on a system-wide basis.

Conference Parties without Public Network Trunks

Enter a number to specify the maximum number of parties allowed in a conference call involving no public network trunks.

Valid entries	Usage
3 to **6**	

Attendant Tone

Valid entries	Usage
y/n	Enter **y** to provide call progress tones to the attendants.

Night Service Disconnect Timer (seconds)

Enter a number or blank to indicate how long a trunk call can be unanswered during night service before being disconnected. The trunk must not have Disconnect Supervision for this timer to apply.

Valid entries	Usage
10 to **1024**	

Bridging Tone

Valid entries	Usage
y/n	Enter **y** to apply a bridging tone when calls are bridged on primary extensions.

Short Interdigit Timer (seconds)

Enter a number to limit the time that digit analysis will wait for the next digit when it has predicted that all the digits have already been collected.

Valid entries	Usage
3 to **9**	

Conference Tone

NOTE:
Bridging and Conference Tones are not supported by all countries. If these tones are enabled for countries other than Italy, Belgium, United Kingdom, or Australia, the tones will be equivalent to no tone (silence) unless the tone is independently administered or customized on the System-Parameters Country Options screen.

Valid entries	Usage
y/n	Enter **y** to provide conference tone as long as three or more calls are in a conference call.

Unanswered DID Call Timer (seconds)

Enter number or blank to limit how long a DID call can remain unanswered before routing to the DID/TIE/ISDN Intercept Treatment feature. This timer interacts with the nonadministrable 50 second Wait for Answer Supervision Timer (WAST). The WAST timer overrides this field. Thus if this field is set to a value equal to or greater than 50 seconds, the caller receives intercept tone instead of the normal attendant or announcement treatment that is given when the Unanswered DID Call Timer expires before the WAST. If the Unanswered DID Call Timer expires while the DID call is being processed by call vectoring, the timer is ignored. See Wait Answer Supervision Timer in this section.

Valid entries	Usage
A number between **10** and **1024**	
blank	A value of blank disables this timer.

Intrusion Tone

Valid entries	Usage
y/n	Enter **y** to apply an intrusion tone (executive override) when an attendant intrudes on the call.

Line Intercept Tone Timer (seconds)

Enter a number to specify how long an analog station user can wait after hearing warning tone without going on hook, before the station is placed in the lockout state.

Valid entries	Usage
2–60	

Special Dial Tone

Special dial tone notifies an analog-phone user if certain features are still active when the user goes off-hook. These features include:

- Call Forwarding
- Send All Calls
- Do Not Disturb

Valid entries	Usage
y/n	Enter **y** to use the Special Dial Tone. You must have a TN2182 circuit pack.

Long Hold Recall Timer (seconds)

You can administer the system to remind a user that a call has been on hold for too long.

Valid entries	Usage
0 - 999	Enter a number between **0** and **999**; **0** deactivates the timer. This value is the number of seconds a call can be on hold before the system re-alerts the user to remind them of the call.

Mode Code Interface

Allows you to use the Mode Code Voice Mail System Interface to connect the switch over a DTMF interface to INTUITY AUDIX or other vendors' voice mail systems.

Reset Shift Timer (seconds)

Used only for station-to-station calls or private network calls using ISDN trunks.

Valid entries	Usage
0 - 255	Specifies the number of seconds that reset shift dial tone is audible before busy tone is heard. Reset shift dial tone allows the user to dial a new extension by dialing one new digit that replaces the last digit of the extension previously dialed. The new digit replaces the last digit of the extension previously dialed. Enter **0** to disable this feature.

Station Call Transfer Recall Timer

Allows a user-transferred call (station-to-station, a trunk call, or a DCS call) to re-terminate with priority ringing back to the station user who initiates the transfer operation if the transfer-to party does not answer the call within the administered Station Call Transfer Recall timer.

Valid entries	Usage
0 - 999	Enter the time in seconds before a call redirects back to the station user who initiated the transfer operation. Enter **0** to disable this feature.

DID Busy Treatment

Valid entries	Usage
attendant	Specifies how to handle a direct inward dialing (DID) call to a busy station.
tone	

Invalid Number Dialed Intercept Treatment

Enter the type of intercept treatment the end-user hears after dialing an invalid number.

Valid entries	Usage
announcement	Provides a recorded announcement when the end-user dials an invalid number. You select and record the message. Enter the extension number for the announcement in the associated field.
tone	Provides intercept tone when the end-user dials an invalid number.

Allow AAR/ARS Access from DID/DIOD

Valid entries	Usage
y/n	Enter **y** to allow calls for DID and DIOD type trunk groups to complete calls using ARS or AAR.

Allow ANI Restriction on AAR/ARS

(For Russia only) If a call is placed over a Russian shuttle trunk or a Russian rotary trunk via an AAR or ARS entry with the ANI Req field set to **r**, then ANI is requested just like a **y** entry. However, if the ANI request fails, the call immediately drops. All other trunk types treat the **r** entry as a **y**.

Valid entries	Usage
y	The ANI Req field on the ''AAR and ARS Digit Analysis Table'' on page 559 and the ''AAR and ARS Digit Conversion Table'' on page 564 permits the additional value of **r**(estricted).
n	The ANI Req field on the two screens takes only the current values of **n** and **y**.

7405ND Numeric Terminal Display

Valid entries	Usage
y/n	If enabled, this allows you to use 7405ND in the Type field of the station screen. This is not an actual phone type, but you can use this to define ports for certain types of Octel Messaging Division voice messaging systems. This numeric display setting sends only numbers, and not names, to the Octel system. You cannot use both 7405ND and 7434ND at the same time.

7434ND

Valid entries	Usage
y/n	If enabled, this allows you to use 7434ND in the Type field of the station screen. This is not an actual phone type, but you can use this to define ports for certain types of Octel Messaging Division systems. You cannot use both 7405ND and 7434ND at the same time. Use this value if your voice messaging system operates in Bridged Mode.

Distinctive Audible Alerting (Internal, External, Priority)

This is also known as Distinctive Ringing.

Valid entries	Usage
1–3.	Enter the number of rings for internal, external and priority calls. For virtual stations, this applies to the mapped-to physical phone.

Attendant Originated Calls

Valid entries	Usage
internal **external** **priority**	Indicates which type of ringing (defined above) applies to attendant-originated calls.

DTMF Tone Feedback Signal to VRU - Connection, Disconnection

This field appears only if DTMF Feedback Signals for VRU on the Customer-Options System Parameters screen is **y**.

Valid entries	Usage
0–9, *, #, ABCD	Enter the code to connect or disconnect the VRU. This can be a single digit, or a combination such as *99 to connect, #99 to disconnect. The tones must be programmed at the VRU as well.
blank	Blank means that no tone is to be sent to the VRU.

Field descriptions for page 6

```
change system-parameters features (page 6)
                          FEATURE-RELATED SYSTEM PARAMETERS

               Abort Transfer? n            No Dial Tone Conferencing? n
        Transfer Upon Hang-Up? n   Select Line Appearance Conferencing? n
Abort Conference Upon Hang-Up? n                                 Unhold? n
   No Hold Conference Timeout: 60



ANALOG BUSY AUTO CALLBACK
                Without Flash?                      Announcement:
                                          Voice Mail Hunt Group Ext:
```

Screen 104. Feature-Related System Parameters screen

Abort Transfer

Stops the transfer operation whenever a user presses a non-idle call appearance button in the middle of the transfer operation, or when they hang up. If both the Abort Transfer and Transfer Upon Hang-Up fields are **y** and you press the TRANSFER button and then dial the complete transfer-to number, hanging up the phone transfers the call. You must select another non-idle call appearance to abort the transfer. If the Transfer Upon Hang-Up field is **y**, hanging up completes the transfer. Requires DCP, Hybrid, IP, ISDN-BRI or wireless phones.

Valid entries	Usage
y/n	Enter **y** to abort the transfer a call by pressing the TRANSFER button, dialing the desired extension, and then hanging up or selecting another non-idle call appearance. The user must press the TRANSFER button again to complete the process unless Transfer Upon Hang-up is also set to **y**.

No Dial Tone Conferencing

When another line is on hold or alerting, No Dial Tone Conferencing, eliminates dial tone while setting up a conference. Refer to GuideBuilder for more information.

Valid entries	Usage
y/n	Enter **y** to activate No Dial Tone Conferencing

Transfer Upon Hang-Up

Allows DCP, hybrid, IP, wireless, or ISDN-BRI phone users to complete a transfer operation by hanging up.

Valid entries	Usage
y/n	Enter **y** so users may transfer a call by pressing the TRANSFER button, dialing the desired extension, and then hanging up. The user may also wait to hang up, speak with the other party, then press TRANSFER again to complete the process.

Select Line Appearance Conferencing

Select Line Appearance Conference changes the capabilities of the conference buttons and line appearance buttons on digital phones. Refer to GuideBuilder for more information.

Valid entries	Usage
y/n	Enter **y** to activate Select Line Appearance Conferencing

Abort Conference Upon Hang-Up

Allows DCP, hybrid, IP, wireless, or ISDN-BRI phone users to abort the conference operation when they hang up.

Valid entries	Usage
y/n	Enter **y** to change a call placed on soft-hold in the conference-pending status to hard-held status if the user hangs up.

Unhold

Allows the user to press the hold button on a phone to release a hold (if no other line appearance is on hold or alerting). This does not apply to BRI phones or attendant consoles.

Valid entries	Usage
y/n	Enter **y** to activate the unhold capability

No Hold Conference Timeout

Controls the timeout of No Hold Conference call setup. The system Answer Supervision timer should be set to a value less than this.

Valid entries	Usage
20-120	Enter the number of seconds.

CPN Replacement for Restricted Calls

Valid entries	Usage
up to 15 characters	Enter a text string to replace the restricted numbers on the display.

CPN Replacement for Unavailable Calls

Valid entries	Usage
up to 15 characters	Enter a text string to replace the unavailable numbers on the display.

Without Flash

Provides automatic callback for analog stations without flashing the hook. It is applied only when the called station is busy and has no other coverage path or call forwarding. The caller can enable the automatic callback without flashing the hook or entering the feature access code.

Valid entries	Usage
y/n	Enter **y** to provide automatic callback for a calling analog station without flashing the hook.

Announcement

Appears only if the Without Flash field is **y**.

Valid entries	Usage
	Enter a valid announcement extension. This field cannot be left blank.

Voice Mail Hunt Group Ext

Appears only if the Without Flash field is **y**.

Valid entries	Usage
y/n	Enter a voice mail hunt group extension.

Field descriptions for page 7

```
change system-parameters features (page 7)
                          FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS

Send Non-ISDN Trunk Group Names as Connected Name?
 Display Connected Name/Number for ISDN DCS Calls?
       Send ISDN Trunk Group Name on Tandem calls?

                              QSIG TSC Extension:
MWI - Number of Digits Per Voice Mail Subscriber:

                             National CPN Prefix:
                        International CPN Prefix:
                       Pass Prefixed CPN to ASAI:
   Unknown Numbers Considered Internal for AUDIX?
            UNSI Calling Name for Outgoing Calls?
              Path Replacement with Measurements?
                 QSIG Path Replacement Extension:
           Path Replace While in Queue/Vectoring?
```

Screen 105. Feature-Related System Parameters screen

Send Non-ISDN Trunk Group Name as Connected Name

Valid entries	Usage
y/n	Enter **y** to send a name of the non-ISDN trunk group as the connected name when a call routes from ISDN to non-ISDN and the call is answered.

Display Connected Name/Number for ISDN DCS Calls

Valid entries	Usage
y/n	Enter **y** to display the connected name/number (if received) for ISDN DCS calls.

Send ISDN Trunk Group Name on Tandem Calls

Valid entries	Usage
y/n	Enter **y** to provide consistent display information regardless of trunk type. If set to **y**, provides only trunk group name.

QSIG TSC Extension

Valid entries	Usage
Enter any valid, unassigned extension.	This is the phantom endpoint extension for QSIG Call Independent Signaling Connections (CISCs), which are similar to NCA Temporary Signaling Connections (TSCs) (both incoming and outgoing).

MWI - Number of Digits Per Voice Mail Subscriber

Appears only if the Basic Supplementary Services field or the ISDN Feature Plus field on the System-Parameters Customer-Options screen is **y**. This field appears an indication of the number of digits per AUDIX subscriber.

NOTE:
For QSIG-MWI only. These routing digits and inserted digits must screen a digit string that, when analyzed and processed, routes to a Signaling Group supporting QSIG-TSCs. Once a QSIG TSC is established (from a message center switch to a served user switch) then every lamp update message places the Inserted Digits field (from the Message Waiting Indication Subscriber Number Prefixes screen) in front of the AUDIX subscriber number to screen a complete QSIG network number for the served user.

NOTE:
For Feature Plus MWI only. The routing digits and inserted digits must screen a digit string that routes over an SSF trunk to the Feature Plus extension on the remote (Served User) switch. The Inserted Digits field must include the Feature Plus extension.

Valid entries	Usage
3 to **7**	Enter a value that corresponds to the digit string length of subscribers translated in the Message Center entity. For instance, if the Message Center entity is AUDIX, the value in this field must match the value of the Extension Length field on the Switch Interface Administration screen of AUDIX.

Feature Plus Ext

Valid entries	Usage
A valid extension	Administration of this field is required for proper termination of some Feature Plus signaling. For example, Message Waiting Indication (MWI) requires this extension in order to send the indication to the appropriate switch. Appears only if the ISDN Feature Plus field is **y** on the System Parameters Customer Options screen.

National CPN Prefix

Allows you to apply prefixes to national calling numbers for display at receiving phones. This is useful for those phones that use or implement call back features based on incoming call numbers. When an ISDN-PRI call arrives, the incoming call setup is analyzed for: (1) whether the Type of Address (TOA) is national or international, and (2) whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony. This administered prefix is applied to national calls. Prefixing applies to any subsequent display on the same switch when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Valid entries	Usage
1 to 5 digits, **0** through **9**, * and **#**	Enter a number that allows you to apply prefixes to national calling numbers for display.

International CPN Prefix

Allows you to apply prefixes to international calling numbers for display at receiving phones. This is useful for those phones that use or implement call back features based on incoming call numbers. When an ISDN-PRI call arrives, the incoming call setup is analyzed for: (1) whether the Type of Address (TOA) is national or international, and (2) whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony. This administered prefix is applied to international calls. Prefixing applies to any subsequent display on the same switch when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Valid entries	Usage
1 to 5 digits, **0** through **9**, * and **#**	Enter a number that allows you to apply prefixes to international calling numbers for display.

Pass Prefixed CPN to ASAI

Passes Calling Party Number information (CPN) to ASAI. The prefixed number is not passed on to other adjuncts, Call Detail Recording, or switches.

Valid entries	Usage
y/n	Enter **y** to pass CPN information to ASAI.

Unknown Numbers Considered Internal for AUDIX

Appears when, on the System-Parameters Customer-Options screen, either the ISDN-PRI or ISDN-BRI Trunks field is **y**. This field controls the treatment of an ISDN number whose numbering plan identification is “unknown” in a QSIG centralized AUDIX arrangement. This field works in conjunction with the Calling Party Number to Intuity AUDIX field on the Hunt Group screen. The Calling Party Number to Intuity AUDIX field on the Hunt Group screen must be **y** for this field to have an effect.

Valid entries	Usage
y	The unknown number is considered “internal” and AUDIX tries to find a calling party name match for the digit string. If a name match is found, AUDIX provides the calling party’s name. If no name is found, AUDIX provides the calling party’s phone number.
n	The unknown number is considered “external” and AUDIX provides the calling party’s phone number.

Maximum Length

Appears only if the Unknown Numbers Considered Internal for AUDIX field is **y**. Indicates the maximum length of an unknown private number. Any unknown number longer than the administered value is considered external. This field cannot be blank when it appears.

Valid entries	Usage
1 - 20	Enter a number for the maximum length of an unknown private number.

USNI Calling Name for Outgoing Calls?

Valid entries	Usage
y/n	Enter **y** to send a name on outgoing calls over NI PRI trunks. **Important:** Be sure you have validated that your service provider's central office is capable of accepting calling name information from DEFINITY in this way. For example, if the central office switch is a 5ESS, it must be a generic 5EXX or later. Failure to validate the central office capability may cause the central office to drop outgoing calls from your switch. In this case, change the value in this field to **n**. Enter **n** to prevent sending calling name information with outgoing calls over NI PRI trunks. **n** is this field overrides a **y** in the Send Name field of the outgoing trunk group screen.

Path Replacement with Measurements

Valid entries	Usage
y/n	Allows QSIG path replacement or DCS with Reroute to be attempted on measured calls.

QSIG Path Replacement Extension

Enter the extension for the system to use as part of the complete number sent in the Path Replacement Propose message.

Valid entries	Usage
Extension	Enter an unused extension that conforms to your dial plan.

Path Replacement While in Queue/Vectoring

Valid entries	Usage
y/n	Enter **y** to allow Path Replacement after queue/vector processing has started. Depending on the version of Call Management System (CMS) you are using, some calls can go unrecorded if you enable this capability. Please see your Avaya representative for more information.

Field descriptions for page 8

```
change system-parameters features (page 8)
                          FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS

    CPN/ANI/ICLID Replacement for Restricted Calls:
   CPN/ANI/ICLID Replacement for Unavailable Calls:
```

Screen 106. Feature-Related System Parameters screen

CPN Replacement for Restricted Calls

Valid entries	Usage
up to 15 characters	Enter a text string to replace the restricted numbers on the display.

CPN Replacement for Unavailable Calls

Valid entries	Usage
up to 15 characters	Enter a text string to replace the unavailable numbers on the display.

Field descriptions for page 9

```
change system-parameters features  (page 9)
                         FEATURE-RELATED SYSTEM PARAMETERS

                 Pull Transfer: n           Update Transferred Ring Pattern? n
         Outpulse Without Tone? y             Wait Answer Supervision Timer? n
         Misoperation Alerting? n              Repetitive Call Waiting Tone? y
    Allow Conference via Flash? y     Repetitive Call Waiting Interval (sec): _
 Vector Disconnect Timer (min): __    Network Feedback During Tone Detection? y
  Hear Zip Tone Following VOA ? y    System Updates Time On Station Displays? n
       Intercept Treatment On Failed Trunk Transfers? n
                     Station Tone Forward Disconnect: silence
                             Level Of Tone Detection: precise
           Charge Display Update Frequency (seconds): 30
         Date Format on 607/2420/4600/6400 Terminals: mm/dd/yy
On-hook Dialing on 607/2420/4600/6400/8400 Terminals? n
RECALL TIMING
        Flashhook Interval? y                  Upper Bound (msec): 1000
                                               Lower Bound (msec): 200
                                  Forward Disconnect Timer (msec): 600
ITALIAN DCS PROTOCOL
  Italian Protocol Enabled? y
   Apply Intercept Locally? _      Enforce PNT-to-PNT Restrictions? _
```

Screen 107. Feature-Related System Parameters screen

Pull Transfer

Valid entries	Usage
y/n	Enter **y** to enable the Pull Transfer feature on a system-wide basis. This allows either the transferring or transferred-to party to press the Transfer button to complete the transfer operation

Update Transferred Ring Pattern

Valid entries	Usage
y/n	Enter **y** to change the ringing pattern from internal to external when an internal station transfers an external call. If most of your calls go through an attendant, you might want to set this to **y**, so your users will be able to distinguish an external call.

Outpulse Without Tone

Valid entries	Usage
y	Enter **y** to indicate the switch will outpulse digits even when a dial tone has not been received.
n	Enter "**n**" if the calling party should receive intercept tone if no dial tone is detected.

Wait Answer Supervision Timer

Refer to Unanswered DID Call Timer (seconds) for more information.

Valid entries	Usage
y	Enter **y** to enable this feature on a systemwide basis. When **y** is entered in this field, calls to stations unanswered after 50 seconds are dropped.
n	When **n** is entered in this field, unanswered calls drop only when the calling party goes on-hook.

Misoperation Alerting

Misoperation Alerting should not be enabled if Call Prompting is optioned.

Valid entries	Usage
y	Enter **y** for misoperation recall alerting on multi-appearance stations, analog stations, and attendant consoles.
n	Enter **n** for standard misoperation handling without recall alerting.

Repetitive Call Waiting Tone

Valid entries	Usage
y/n	Enter **y** to indicate that a repetitive call waiting tone be provided to the called party for all types of call waiting access.

Allow Conference via Flash

Valid entries	Usage
y	Enter **y** to allow an analog station to use flash to conference calls together.
n	Enter **n** to prevent this.

Repetitive Call Waiting Interval (sec)

Valid entries	Usage
1 to **99** in increments of 1	Enter a number to specify the number of seconds between call waiting tones. This field appears when the Repetitive Call Waiting Tone is **y**.

Vector Disconnect Timer (min)

Enter the number of minutes, or blank that a trunk should remain connected to a vector.

Valid entries	Usage
1 to **240**	The number of minutes that you enter determines when the trunk will be disconnected if the Disconnect Supervision-In or Disconnect Supervision-Out fields on the Trunk screen are **n**.
blank	Enter blank if you do not want Avaya MultiVantage to initiate a disconnect.

Network Feedback During Tone Detection

Valid entries	Usage
y/n	Enter **y** to provide audible feedback to the user while the system attempts to detect dial tone.

Hear Zip Tone Following VOA?

This tone alerts a telephone user that the announcement has completed and a caller is now connected. CallMaster set and attendant console users hear double zip tone following the announcement. All other telephone users hear single zip tone.

NOTE:
This field does not effect auto-answer zip tone heard prior to the VOA.

Valid entries	Usage
y	Enter **y** to play zip tone following a VDN of Origin Announcement (VOA).
n	Enter **n** if you do not want zip tone following a VOA.

System Updates Time On Station Displays

This does not apply to phones (such as BRI phones) where the user sets the time.

Valid entries	Usage
y/n	Enter **y** to have the system automatically update the time on display phones when background maintenance is run (for example, when the set is plugged in).

Intercept Treatment on Failed Trunk Transfers

Valid entries	Usage
y	Enter **y** to provide intercept treatment to calls failing trunk transfers.
n	Enter **n** to drop these calls.

Station Tone Forward Disconnect

Tone Forward Disconnect applies to any station other than one administered as a data endpoint, an attendant console, a BRI phone, an auto answer, or as an Outgoing Call Management (OCM) agent.

Valid entries	Usage
busy **intercept** **silence**	When a station is the last party remaining off-hook on a call, that station receives the indicated tone or silence until that station is placed on-hook, or until the tone has played for 45 seconds and is followed by silence.

Level of Tone Detection

For the most part, this option is no longer required in today's switching environment. It may be useful if your users are having difficulty placing outgoing calls due to inaccurate detection of network dial tone.

Valid entries	Usage
broadband	This is the least exact of the levels of tone detection. If the switch detects any tone at all, it interprets this as dial tone.
medium	The switch interprets any tone which has a continuous "on" period of longer than 1 second as dial tone. Otherwise, the switch accepts whatever the tone detector circuit pack reports.
precise	The switch accepts whatever the tone detector circuit pack reports.

Charge Display Update Frequency (seconds)

This applies only if you use Advice of Charge or Periodic Pulse Metering with display functions.

Valid entries	Usage
10–60	The amount of time (in seconds) between charge-display updates. Frequent display updates may have considerable performance impact. If the duration of a call is less than the Charge Display Update Frequency, the display will not automatically show charge information. To see charge information for a call, the user must have a disp-chrg button and must press the button before the call drops.

Date Format on 607/2420/4600/6400 Terminals

The format of the date as displayed on the telephones.

Valid entries	Usage
mm/dd/yy	month/day/year
dd/mm/yy	day/month/year
yy/mm/dd	year/month/day

On-hook Dialing on 607/2420/4600/6400/8400 Terminals

For 6400/8400, 607, 2420, and 4600 telephone users with speakerphones.

Valid entries	Usage
y/n	Enter **y** allows users to dial without lifting the handset. If you enable this, users hear dial tone when they press the Speaker button, even if the handset is on-hook.

The next four fields control station-to-switch recall signal timing. If a flashhook interval (recall window) is required, the upper and lower bounds of the interval can be administered. An on-hook that lasts for a period of time greater than or equal to the lower bound and less than or equal to the upper bound will be treated as a recall flash. If an interval is not required, the Disconnect Timing value must be administered. An on-hook that lasts for a period of time less than this value will be ignored; greater than or equal to this value will be regarded as a disconnect. Regardless, an on-hook lasting 50 to 150 ms coming from a 2500-type set will always be treated as a digit pulse unless Ignore Rotary Digits is **y** for that station.

Flashhook Interval

Valid entries	Usage
y	Enter **y** to indicate that a flashhook interval (recall window) is required. If a **y** is entered, Upper Bound and Lower Bound appear
n	If **n** is entered, Disconnect Timing appears.

Upper Bound (msec)

Specifies the upper bound of the flashhook interval. Specifies the upper bound of the station-to-switch recall signal timing interval in milliseconds. Appears when Flashhook Interval is **y**.

Valid entries	Usage
80 to **1250** (in increments of 10).	A flash of 50 msec to 130 msec is always acceptable from a 2500-type set regardless of the setting of the Upper and Lower bounds and will be treated as the digit one.

Lower Bound (msec)

The lower bound of the station-to-switch recall signal timing interval in milliseconds. Specifies the lower bound of the flashhook interval. Appears when Flashhook Interval is **y**.

Valid entries	Usage
80 to **1250** (in increments of 10).	

Forward Disconnect Timer (msec)

Specify the duration of a momentary disconnect sent by the switch to an analog station user when that user is the last party still off-hook on a call.

Valid entries	Usage
25 to **1500** (in increments of 25).	

The next three fields control the Italian DCS Protocol feature.

Italian Protocol Enabled

Valid entries	Usage
y/n	Enter **y** to enable the Italian DCS feature on a systemwide basis.

Apply Intercept Locally

This field appears only if Italian Protocol Enabled is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that DID/CO intercept treatment will be applied locally instead of on the originating switch.

Enforce PNT-to-PNT Restrictions

This field appears only if Italian Protocol Enabled is **y.**

Valid entries	Usage
y/n	Enter **y** to indicate that restrictions and denial of PNT-to-PNT connections will be enforced when the EDCS message is unavailable. A **y** in this field means restrictions will be enforced.

Field descriptions for page 10

```
change system-parameters features (page 10)
                        FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER SYSTEM PARAMETERS
  EAS
        Expert Agent Selection (EAS) Enabled? n
       Minimum Agent-LoginID Password Length:
         Direct Agent Announcement Extension: ________          Delay: ___
   Message Waiting Lamp Indicates Status For: station

VECTORING
                   Converse First Data Delay: 0       Second Data Delay: 2
              Converse Signaling Tone (msec): 100          Pause (msec): 70_
                    Prompting Timeout (secs): 10
                Interflow-qpos EWT Threshold: 2
   Reverse Star/Pound Digit For Collect Step? n
         Available Agent Adjustments for BSR? _
SERVICE OBSERVING
             Service Observing: Warning Tone? n      or Conference Tone?
ASAI
Call Classification After Answer Supervision? n      Send UCID to ASAI? n
```

Screen 108. Feature-Related System Parameters screen

Call Center System Parameters:

Expert Agent Selection (EAS) Enabled

To enable this field, either no ACD or vectoring hunt groups may exist or, existing ACD or vectoring hunt groups must be “skilled.” Only appears if Expert Agent Selection (EAS) on the System-Parameters Customer-Options screen is **y**.

Valid entries	Usage
y/n	Enter **y** to enable Expert Agent Selection.

Minimum Agent-LoginID Password Length

Enter the minimum number of digits that must be administered as an EAS Agent’s LoginID password. Only appears if Expert Agent Selection (EAS) on the System-Parameters Customer-Options screen is **y**.

Valid entries	Usage
0–**9**	Entering a **0** or blank indicates no password is required.

Direct Agent Announcement Extension

Valid entries	Usage
Valid extension	Enter the extension of the direct agent announcement.

Direct Agent Announcement Delay

Only appears if Expert Agent Selection (EAS) or ASAI Link Core Capabilities on the System-Parameters Customer-Options screen is **y**.

Valid entries	Usage
0–**99**	Enter the number of seconds the caller will hear ringback before the Direct Agent Announcement is heard by the calling party.

Message Waiting Lamp Indicates Status For

Only appears if Expert Agent Selection (EAS) on the System-Parameters Customer-Options screen is **y**.

Valid entries	Usage
station	Since you only have one message waiting lamp on a phone, you need to indicate if the message is for at the phone extension or the loginID.
loginID	Expert Agent Selection (EAS) must be enabled to use this option.

Converse First Data Delay/Second Data Delay

The First Data Delay prevents data from being outpulsed (as a result of a converse vector step) from the system to CONVERSANT before CONVERSANT is ready. The delay commences when the CONVERSANT port answers the call. The Second Data Delay is used when two groups of digits are being outpulsed (as a result of a converse vector step) from the system to CONVERSANT. The Second Data Delay prevents the second set from being outpulsed before CONVERSANT is ready. The delay commences when the first group of digits has been outpulsed. Only appears if Vectoring (Basic) on the System-Parameters Customer-Options screen is **y**.

Valid entries	Usage
0 to **9**	Number of seconds for the delay.

Converse Signaling Tone/Pause

Only appears if Vectoring (Basic) and DTMF on the System-Parameters Customer-Options screen are **y**. In the Signaling Tone field, enter the length in milliseconds of the digit tone for digits being passed to the CONVERSANT. In the Pause field, enter the length in milliseconds of the delay between digits being passed. The optimum timers for the CONVERSANT are a 100 msec tone and 70 msec pause.

Valid entries	Usage
40 to **2550** (in increments of 10).	Values entered in the Tone/Pause fields are rounded up or down depending upon the type of circuit pack used to outpulse the digits.
100	■ **TN742B or later suffix analog board** — Tone and pause round up or down to the nearest 25 msecs. For example a 130 msec tone rounds down to 125 msecs, a 70 msec pause rounds up to 75 msec for a total of 200 msecs per tone. ■ **TN464F, TN767E or later suffix DS1 boards** — Tone and pause round up to the nearest 20 msecs. For example a 130 msec tone rounds up to 140 msecs, a 70 msec pause rounds up to 80 msecs for a total of 220 msecs per tone. If a circuit pack has been used for end-to-end signalling to the CONVERSANT, and has then been used to send digits to a different destination, the CONVERSANT timers may stay in effect. To reset your timers to the system default, pull and reseat the circuit pack.

Prompting Timeout (secs)

Enter the number of seconds before the Collect Digits command times out for callers using rotary dialing. Only appears if Vectoring (Prompting) on the System-Parameters Customer-Options screen is **y**.

Valid entries	Usage
4 to **10**	

Interflow-qpos EWT Threshold

Part of enhanced Look-Ahead Interflow. Any calls predicted to be answered before this threshold will not be interflowed (therefore saving CPU resources).

Valid entries	Usage
0–**9**	Number of seconds for this threshold

Available Agent Adjustments for BSR

Controls the use of BSR available agent adjustments. BSR must be **y** on the Feature-Related System-Parameters screen.

Valid entries	Usage
y/n	Enter **y** to allow adjustments to available agents.

Service Observing Warning Tone

Service Observing (Basic) on the System-Parameters Customer-Options screen must be **y** before this field may be administered.

CAUTION:

The use of Service Observing features may be subject to federal, state, or local laws, rules or regulations or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves and comply with all applicable laws, rules, and regulations before using these features.

Valid entries	Usage
y/n	Enter **y** to assign a warning tone to be given to telephone users and calling parties whenever their calls are being monitored using the Service Observing feature.

Call Classification After Answer Supervision?

For use with ASAI Outbound Call Management (OCM).

Valid entries	Usage
y/n	Enter **y** to force the switch to rely on the network to provide answer/busy/drop classification to the switch. After the call has been answered, a call classifier can be added to perform answering machine, modem and voice answering detection.

Send UCID to ASAI

Valid entries	Usage
y/n	Enter **y** to enable transmission of Universal Call ID (UCID) information to ASAI.

Field descriptions for page 11

```
change system-parameters features  (page 11)
                        FEATURE-RELATED SYSTEM PARAMETERS

AGENT AND CALL SELECTION
                     MIA Across Splits or Skills? n
                      ACW Agents Considered Idle? y
                      Call Selection Measurement: current-wait-time
Service Level Supervisor Call Selection Override? y
                             Auto Reserve Agents:

REASON CODES
                       Aux Work Reason Code Type: none
                         Logout Reason Code Type: none
CALL MANAGEMENT SYSTEM
                             Adjunct CMS Release:
                 ACD Login Identification Length: 0
               BCMS/VuStats Measurement Interval: hour
       BCMS/VuStats Abandon Call Timer (seconds):
                 Validate BCMS/VuStats Login IDs? n
                        Clear VuStats Shift Data: on-login
             Remove Inactive BCMS/VuStats Agents? n
```

Screen 109. Feature-Related System Parameters screen

Agent and Call Selection Parameters:

MIA Across Splits or Skills?

Valid entries	Usage
y/n	Enter **y** to remove an agent from the MIA queue for all the splits/skills/hunt groups that he or she is available in when the agent answers a call from any of his or her splits/skills/hunt groups.

ACW Agents Considered Idle

Valid entries	Usage
y/n	Enter **y** to have agents who are in After Call Work included in the Most-Idle Agent queue. This means that ACW is counted as idle time. Enter **n** to exclude ACW agents from the queue.

Call Selection Measurement

This field determines how Avaya MultiVantage selects a call for an agent when the agent becomes available and there are calls in queue.

For information on CentreVu® Advocate, please contact your Avaya Account Executive or see the *Avaya Business Advocate User Guide*.

Valid entries	Usage
current-wait-time	Current Wait Time selects the oldest call waiting for any of the agent's skills.
predicted-wait-time.	Predicted Wait Time is a feature of CentreVu® Advocate.

Service Level Supervisor Call Selection Override

This field determines whether Avaya MultiVantage changes agents' call handling preferences when a skill using Service Level Supervisor exceeds its Level 1 threshold.

For information on CentreVu® Advocate, please contact your Avaya Account Executive or see the *Avaya Business Advocate User Guide*.

Valid entries	Usage
y	Enter **y** if you want to override the normal call handling preferences of a skill's assigned agents in this situation.
n	Enter **n** if you do not want to override agents' normal call handling preferences when the skill exceeds its Level 1 threshold. Service Level Supervisor requires Expert Agent Selection and CentreVu® Advocate.

Auto Reserve Agents

Valid entries	Usage
all	
none	
secondary-only	

Reason Codes Parameters

Aux Work Reason Code Type

Valid entries	Usage
none	Enter **none** if you do not want an agent to enter a Reason Code when entering AUX work.
requested	Enter **requested** if you want an agent to enter a Reason Code when entering AUX mode but do not want to force the agent to do so. To enter **requested** the Reason Codes and EAS on the System-Parameters Customer-Option screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the Reason Codes and EAS on the System-Parameters Customer-Option screen must be **y**.

Logout Reason Code Type

Valid entries	Usage
none	Enter **none** if you do not want an agent to enter a Reason Code when logging out.
requested	Enter **requested** if you want an agent to enter a Reason Code when logging out but do not want to force the agent to do so. To enter **requested** the Reason Codes and EAS on the System-Parameters Customer-Option screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when logging out. Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the Reason Codes and EAS on the System-Parameters Customer-Option screen must be **y**.

Call Management System Parameters:

Adjunct CMS Release

Specifies the release of the CMS adjunct used with the system.

Valid entries	Usage
R2 **R3** **R3V2** **R3V4** **R3V5** **R3V6** blank	For CMS, this field cannot be blank.

ACD Login Identification Length

Enter the number of digits for an ACD Agent Login ID if Expert Agent Selection (EAS) on the System-Parameters Customer-Options screen is **n**. If BCMS/VuStats Login IDs is **y**, the ACD Login ID length must be greater than 0. This field identifies an ACD agent to CMS. The number you enter in this field must equal the number of characters in the agent’s login ID.

Valid entries	Usage
0–**9**	For CMS, this field cannot be 0.

BCMS/VuStats Measurement Interval

You can enter **half-hour** or **hour** for polling and reporting measurement data if the BCMS (Basic) and/or the VuStats on the System-Parameters Customer-Options screen is **y**.

Valid entries	Usage
half-hour **hour**	There are a maximum of 25 time slots available for measurement intervals. If **hour** is specified, an entire day of traffic information will be available for history reports; otherwise, only half a day will be available. This does not affect daily summaries as they always reflect traffic information for the entire day. The interval may be changed at any time, but will not go into effect until the current interval completes.

BCMS/VuStats Abandon Call Timer (seconds)

Specifies the number of seconds before calls are considered abandoned. Calls with talk time that is less than this number (and that are not held) are tracked by BCMS and displayed by VuStats as ABAND calls.

Valid entries	Usage
1–10	

Validate BCMS/VuStats Login IDs

Valid entries	Usage
y	Enter **y** to allow entry only of login-IDs that have been entered on the BCMS Login-ID screen.
n	Enter **n** to allow entry of any ACD login of the proper length.

Clear VuStats Shift Data

Valid entries	Usage
on-login	Enter **on-login** to clear shift data for an agent when the agent logs in.
at-midnight	Enter **at-midnight** to clear shift data for all agents at midnight.

Remove Inactive BCMS/VuStats Agents

If option **yes** is used, agents are removed from reports when they have no staff time during the previous 7 days. If option **no** is used, the agents remain on the report even if they have no staff time for any period of time.

Field descriptions for page 12

```
change system-parameters features  (page 12)
                        FEATURE-RELATED SYSTEM PARAMETERS

AUTOMATIC EXCLUSION PARAMETERS

                       Automatic Exclusion by COS? y
                Automatic Exclusion Coverage/Hold? y
            Automatic Exclusion with Whisper Page? y
                              Recall Rotary Digit: 2
                    Password to Change COR by FAC: *
         Duration of Call Timer Display (seconds): 3
WIRELESS PARAMETERS
  Radio Controllers with Download Server Permission (enter board location)
    1.         2.         3.         4.         5.

IP PARAMETERS
                   Direct IP-IP Audio Connections? n
                             IP Audio Hairpinning? n
RUSSIAN MULTI-FREQUENCY PACKET SIGNALING
                                          Re-try?
     T2 (Backward Signal) Activation Timer (secs):
```

Screen 110. Feature-Related System Parameters screen

Automatic Exclusion by COS

Activates automatic exclusion automatically by class of service when a user goes off-hook on a station with an assigned EXCLUSION button. This works only for stations on the local switch.

Valid entries	Usage
y	Enables automatic exclusion by a class of service.
n	Exclusion operates normally. See Exclusion on ''Telephone feature buttons'' on page 129 for more information.

Automatic Exclusion Coverage/Hold

Appears when Automatic Exclusion by COS field is **y**.

Valid entries	Usage
y	The principal can bridge onto the call by pressing the appropriate bridged appearance button. And, if the coverage point places the exclusion call on hold, the principal can retrieve the call.
n	If a coverage point has answered a call and there is active exclusion on the call, the principal cannot bridge onto the call. And, if the coverage point places the exclusion call on hold, the principal cannot retrieve the call.

Automatic Exclusion with Whisper Page

Appears when Automatic Exclusion by COS field is **y**.

Valid entries	Usage
y	The whisper page goes through to an excluded call.
n	The whisper page is denied when a station attempts to whisper page to a station that is on an excluded call.

Recall Rotary Digit

This establishes the digit to use for rotary phones to receive recall dial tone. Dialing this digit simulates switch hook flash so that users of rotary phones can use features such as conference and transfer. The phone must also be administered to use the recall rotary digit.

Valid entries	Usage
0–9	Enter the digit users can dial to generate recall dial tone. Use a number that is not the first digit in normal dialing patterns.

Password to Change COR by FAC

Appears if, on the System-Parameters Customer-Options screen, the Change COR by FAC field is **y**. Avaya recommends using this password option.

Valid entries	Usage
4 - 8 digits	Requires the password option.
blank	Disables the password option.

Duration of Call Timer Display

Administer a call timer button on the Station screen.

Valid entries	Usage
3–30	Enter the length of time (in 3 second increments) that the call information remains on display after the call is terminated.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints

Valid entries	Usage
y/n	Enter to **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack on the switch.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack on the switch in IP format, without going through the DEFINITY TDM bus.

Re-try

The Re-try field applies to outgoing Russian MFP trunks. It allows the switch to resend Russian MFP calling party number and dialed number information to the CO. The switch resends the information only once over another outgoing trunk port of the same trunk group if the switch receives a message that the information was received incorrectly by the CO. The switch also sends Russian MFP information over another trunk port if the switch does not receive a timely response for the information.

Valid entries	Usage
y/n	Enter **y** to resend address information on outgoing Russian MFP trunks.

T2 (Backward Signal) Activation Timer (secs)

The T2 (Backward Signal) Activation Timer (secs) field applies to outgoing Russian MFP trunks. This field sets the number of seconds the switch waits for confirmation after sending calling party number and dialed number information on outgoing Russian MFP trunks

Valid entries	Usage
5 - 20	Enter the number of seconds the system waits to receive confirmation after sending the address information on outgoing Russian MFP trunks.

Radio Controllers with Download Server Permission

Enter the port location of the circuit pack.

Valid entries	Usage
	Enter location of the circuit pack containing the radio controllers with download server permission.

Group Paging Using Speakerphone

Use this screen to assign digital speakerphones to a paging group. Users can page all the phones in the group simultaneously by dialing the group's extension.

```
add group-page next                                            Page 1 of 1
                    GROUP PAGING USING SPEAKERPHONE
   Group Number: 1                             Group Extension: 3210
     Group Name: Sales staff                               COR: 5
GROUP MEMBER ASSIGNMENTS
   Ext      Name                                  Ext      Name
 1: 2009    B. Smith                           17:
 2: 2010    R. Munoz                           18:
 3: 2011    Y. Lu                              19:
 4: 2012    A. Sullivan                        20:
 5:                                            21:
 6:                                            22:
 7:                                            23:
 8:                                            24:
 9:                                            25:
10:                                            26:
11:                                            27:
12:                                            28:
13:                                            29:
14:                                            30:
15:                                            31:
16:                                            32:
```

Screen 111. Group Paging Using Speakerphone screen

Group Number

This field displays the identifying number the switch assigns to the group when it is created.

Group Extension

Valid entries	Usage
An extension number	Assign the extension users will dial to page the members of this group.

Group Name

Valid entries	Usage
1 to 27 characters	Enter a name that's informative to users, because it appears on callers' phone displays when they page the group.

COR

Valid entries	Usage
0 to **95**	Enter a class of restriction. In order to page the group, users' class of restriction must give them calling permission for the group's class of restriction.

Ext

Valid entries	Usage
An extension number	Assign a phone to the group by entering its extension number in this field.

Name

When you save your changes, the switch fills in this display field with the name assigned to each extension on the Station screen.

Related topics

Refer to ''Paging over speakerphones'' on page 469 for complete instructions.

Refer to ''Group paging'' on page 1570 for a description of the feature.

Holiday Table

Use this screen to define individual holidays or holiday ranges for use with Holiday Vectoring. To use and administer Holiday Vectoring, Vectoring (Holiday) must be **y** on the System-Parameters Customer-Options screen. Contact your Avaya representative for assistance.

```
 change holiday table 1                                              Page   1 of 1
                                   HOLIDAY TABLE

Number:  1                      Name:

______START______       ______END________
Month Day Hour Min      Month Day Hour Min      Description
__    __  __   __       __    __  __   __       ___________________________________
__    __  __   __       __    __  __   __       ___________________________________
__    __  __   __       __    __  __   __       ___________________________________
__    __  __   __       __    __  __   __       ___________________________________
__    __  __   __       __    __  __   __       ___________________________________
__    __  __   __       __    __  __   __       ___________________________________
__    __  __   __       __    __  __   __       ___________________________________
__    __  __   __       __    __  __   __       ___________________________________
```

Screen 112. Holiday table screen

Number

Display only field identifying the holiday table number.

Valid display	Usage
1-10	Holiday table number.

Name

Valid entries	Usage
Up to 27 characters	Enter a description of the holiday table.

Start Month

Valid entries	Usage
1-12	Enter the starting month of the holiday.

Start Day

Valid entries	Usage
1-31	Enter the starting day of the holiday.

Start Hour

Valid entries	Usage
0-23	Enter the starting hour of the holiday using a 24-hour clock.

Start Min

Valid entries	Usage
0-59	Enter the starting minute of the holiday.

End Month

Valid entries	Usage
1-12	Enter the ending month of the holiday.

End Day

Valid entries	Usage
1-31	Enter the ending day of the holiday.

End Hour

Valid entries	Usage
0-23	Enter the ending hour of the holiday using a 24-hour clock.

End Min

Valid entries	Usage
0-59	Enter the ending minute of the holiday.

Description

Valid entries	Usage
Up to 27 characters.	Enter a phrase to describe the holiday.

Hospitality

This screen is used to implement the system parameters associated with the hospitality features. To use and administer the Hospitality-related features, Hospitality must be y on the System-Parameters Customer-Options screen. Contact your Avaya representative for assistance.

Field descriptions for page 1

```
change system-parameters hospitality                            Page   1 of   3
                             HOSPITALITY

                      Message Waiting Configuration: act-nopms
              Controlled Restrictions Configuration: act-nopms
              Housekeeper Information Configuration: act-nopms
                    Number of Housekeeper ID Digits: 0
                                   PMS Log Endpoint:
                          Journal/Schedule Endpoint:
            Client Room Coverage Path Configuration: act-nopms
             Default Coverage Path for Client Rooms:
            Forward PMS Messages to Intuity Lodging? n

                             PMS LINK PARAMETERS
                                       PMS Endpoint:
                                  PMS Protocol Mode: transparent ASCII mode? n
               Seconds before PMS Link Idle Timeout: 20
Milliseconds before PMS Link Acknowledgment Timeout: 500
                     PMS Link Maximum Retransmissions: 3
           PMS Link Maximum Retransmission Requests: 3
                     Take Down Link for Lost Messages? y
```

Screen 113. Hospitality screen

Message Waiting Configuration

This indicates whether message waiting notification requests and changes are being exchanged between the server and the PMS.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	Message waiting is active on the server and information between the PMS and server is being transmitted.

Controlled Restrictions Configuration

This indicates whether controlled restriction information is being exchanged between the server and the PMS.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	The server and the PMS exchange and accept controlled restriction information.

Housekeeper Information Configuration

This indicates whether housekeeper information is being exchanged between the server and the PMS.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	If active (**act-pms**), the server and PMS exchange and accept housekeeper information.

Number of Housekeeper ID Digits

Valid entries	Usage
0 to **6**	Enter the number of digits that the housekeeper must dial for identification.

PMS Log Endpoint

This is a valid data extension number that is assigned to the data module connected to the PMS/Log printer.

Valid entries	Usage
Valid data extension	Cannot be a VDN extension. This extension is dialed by the server to send housekeeping and PMS events to the printer.
PMS_LOG	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_LOG on the IP Services screen.
PMS_JOURNAL	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_JOURNAL on the IP Services screen.

Journal/Schedule Endpoint

This is a valid data extension number that is assigned to the data module connected to the Journal/Schedule printer.

Valid entries	Usage
Valid data extension number	Cannot be a VDN extension. This extension can be the same as the PMS/Log printer and both sets of reports can be printed on the same printer. This extension is dialed by the server to send journal information or schedule reports to the printer.
PMS_LOG	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_LOG on the IP Services screen.
PMS_JOURNAL	Use this value if the printer is connected over a TCP/IP link, and this link is defined as PMS_JOURNAL on the IP Services screen.

Client Room Coverage Path Configuration

This indicates whether the server and the PMS exchange coverage path information for guest stations.

Valid entries	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	If active (**act-pms**), the server and PMS exchange and accept coverage path information. This field does not apply to normal mode. When upgrading from a release that does not support this feature, the field is set to **act-pms** if the PMS protocol mode is administered for transparent or ASCII mode.

Default Coverage Path for Client Rooms

This applies only to stations with a “client room” class of service in the “occupied” mode. This field is used for transparent or ASCII mode. The value in this field is also used during a translation save as the coverage path for each station with “client room” class of service.

Valid entries	Usage
1 to **999**	Enter the coverage path assigned when the server receives a check-out message for a valid extension or a new check-in.

Forward PMS Message to INTUITY Lodging

This field is used only in ASCII mode.

Valid entries	Usage
y	PMS-to-INTUITY messages are sent through the server.
n	PMS-to-INTUITY messages are sent directly to the Avaya INTUITY Lodging system.

PMS Log Endpoint

Valid entries	Usage
Valid extension	Enter the data extension number the server dials to access PMS. Cannot be a VDN extension.
PMS	Use this value if the PMS is connected over a TCP/IP link, and this link is defined as PMS on the IP Services screen.

PMS Protocol Mode

This indicates the message protocol mode used between the server and PMS. Coordinate this option with your PMS vendor.

Valid entries	Usage
normal	
transparent	

ASCII mode

The PMS Protocol Mode field must be **transparent**.

Valid entries	Usage
y/n	Enter **y** when the ASCII-only mode is being used for the PMS message set.

Seconds Before PMS Link Idle Timeout

Valid entries	Usage
5 to **20**	Enter the idle time in seconds that the server waits for an acknowledgment from the PMS before the server enters link failure mode from the PMS transmission link.

Milliseconds Before PMS Link Acknowledgment Timeout

This regulates how quickly the system responds to a message from the PMS (also known as “pace timing.”) This value is also used as the “inquiry message” (ENQ) time-out value. In most cases, keep this as short as possible.

Valid entries	Usage
100 to **2000**	Enter the time in milliseconds the system waits for an acknowledgment from the PMS indicating it correctly received a message.

PMS Link Maximum Retransmissions

Valid entries	Usage
1 to **5**	Enter the number of times that the server retransmits a message to the PMS in response to a negative acknowledgment, or sends an inquiry for acknowledgment from the PMS before giving up on the message.

PMS Link Maximum Retransmission Requests

Valid entries	Usage
1 to **5**	Enter the number of times that the server will allow the PMS to request acknowledgment for a message that it sent.

Take Down Link for Lost Messages

Valid entries	Usage
y/n	Enter **y** to cause the PMS link to come down if messages are being lost. Monitor your PMS error log if you use n.

Field descriptions for page 2

```
                                                     Page   2 of   3
                                  HOSPITALITY
              Dual Wakeup? y    Daily Wakeup? y  VIP Wakeup? y
                                       VIP Wakeup Per 5 Minutes: _____
                               Room Activated Wakeup With Tones?
                       Time of Scheduled Wakeup Activity Report: ______
                        Time of Scheduled Wakeup Summary Report: ______
             Time of Scheduled Emergency Access Summary Report: ______
                                              Announcement Type:

          Length of Time To Remain Connected To Announcement: ____
             Extension To Receive Failed Wakeup LWC Messages: ______
           Routing Extension On Unavailable Voice Synthesis: ______
                       Display Room Information in Call Display?
                             Automatic Selection of DID Numbers?
                            Custom Selection of VIP DID Numbers?
                                      Number of Digits from PMS:
                                               PMS Sends Prefix?
                          Number of Digits in PMS Coverage Path:
                                         Digit to Insert/Delete:
```

Screen 114. Hospitality screen

Dual Wakeup

Valid entries	Usage
y/n	Enter **y** if each extension can request two wakeup calls within one 24-hour time period.

Daily Wakeup

Valid entries	Usage
y/n	Enter **y** if each extension can request daily wakeup calls.

VIP Wakeup

Valid entries	Usage
y/n	Enter **y** if each extension can request VIP wakeup calls.

VIP Wakeups Per 5 Minutes

This field appears only if VIP Wakeup is **y**.

Valid entries	Usage
1 through 50	Enter the number of VIP Wakeup calls allowed in a 5-minute interval.

Room Activated Wakeup with Tones

Valid entries	Usage
y/n	Enter **y** if wakeup calls can be activated via tones that prompt users for the time they wish to waken. This allows room activated wakeup calls without the use of a speech synthesizer or a display telephone.

CAUTION:

Set the following reports for a time other than when the system does its scheduled maintenance tests. To make sure the times do not overlap, enter the command display system-parameters maintenance and check when the system is set to run tests.

Time of Scheduled Wakeup Activity Report

This indicates the time of day that the Wakeup Activity Report will be printed on the Journal/Schedule Printer. This report summarizes the wakeup activity for each extension that had wakeup activity for the past 24 hours.

Valid entries	Usage
hh:mm:am/pm	Enter the time where hh=hour, mm=minute, am/pm=A.M. or P.M.

Time of Scheduled Wakeup Summary Report

This indicates the time of day that the Wakeup Summary Report will be printed on the Journal/Schedule printer. This report gives an hour-by-hour summary of the number of scheduled wakeup calls and a list of extensions to which wakeup calls were attempted but did not complete during the hour.

Valid entries	Usage
hh:mm:am/pm	Enter the time where hh=hour, mm=minute, am/pm=A.M. or P.M.

Time of Scheduled Emergency Access Summary Report

This indicates the time of day that the Emergency Access Summary Report will be printed on the Journal/ Schedule printer.

Valid entries	Usage
hh:mm:am/pm	Enter the time where hh=hour, mm=minute, am/pm=A.M. or P.M.

Announcement Type

This indicates the type of automatic wakeup announcement the hotel guest will receive. Allowable entries are as follows:

Valid entries	Usage
external	Applicable when using an announcement adjunct. If **external** is used, complete the **Auxiliary Board for Announcement** field.
integrated	Applicable when using the TN750B or TN750C announcement circuit pack. If **integrated** is used, complete the **Integrated Announcement Extension** field. The extension you enter must be a valid integrated announcement extension (administered on the Recorded Announcements screen) or a VDN. If you enter an invalid extension, the server displays an error message.
mult-integ	Multi-integrated; applicable when using the TN750B or TN750C announcement circuit pack. If **mult-integ** is used, complete the **Default Announcement Extension** field. The extension you enter must be a valid integrated announcement extension (administered on the Recorded Announcements screen) or a VDN. If you enter an invalid extension, the server displays an error message.
voice-synthesis	If **voice-synthesis** is used, complete the **Announcement Ports** field.
music-on-hold	If **music-on-hold** is used, no other field appears.
silence	If **silence** is used, no other field appears.

NOTE:
One of the following four fields appears depending on what data is entered in the Announcement Type field.

Auxiliary Board for Announcement

This field appears only when the **external** announcement type is used. This indicates the equipment location of an auxiliary trunk circuit that connects to the external announcement equipment. Enter a 5-character circuit pack number.

Valid entries	Usage
1 to **3**	cabinet
A to **E**	carrier
0 to **20**	slot

Integrated Announcement Extension

This field appears only when the **integrated** announcement type is used. This indicates the wakeup announcement extension when using the integrated announcement circuit pack.

Valid entries	Usage
valid extension or VDN	Enter the extension of the announcement you want to use for wakeup calls.

Default Announcement Extension

This field appears only when the **mult-integ** announcement type is used. This indicates the default wakeup announcement extension when using the integrated announcement circuit pack.

Valid entries	Usage
valid extension or VDN	Enter the extension of the announcement you want to use for default wakeup calls.

Announcement Ports

This field appears only when the **voice-synthesis** announcement type is used. For the **voice-synthesis** announcement type, this indicates the equipment location of two ports on the voice synthesizer circuit pack.

Valid entries	Usage
1 to **3**	cabinet
A to **E**	carrier
0 to **20**	slot
01 to **04**	circuit

Length of Time to Remain Connected to Announcement

Enter the length of time in seconds that a hotel guest will be connected to an announcement. This applies only after the guest has heard the announcement completely one time, but continues to listen.

Valid entries	Usage
0 to **300**	

Extension to Receive Failed Wakeup LWC Messages

This indicates where unsuccessful wakeup LWC messages will be stored. This is usually administered to an unassigned extension (cannot be a VDN extension) or to the attendant (attd). In addition, a LWC lamp for that extension is usually assigned to the attendant console as an indication of failed wakeup calls.

Valid entries	Usage
Unassigned extension or **attd**	

Routing Extension on Unavailable Voice Synthesis

This indicates where a wakeup call will go to if both wakeup announcements on the Speech Synthesizer circuit pack are not available. This is usually administered to an unassigned extension (cannot be a VDN extension) or to the attendant (attd).

Valid entries	Usage
Assigned extension or **attd**	

Display Room Information in Call Display

This indicates the type of guest room information displayed on phone displays.

Valid entries	Usage
y	If this field is set to **y**, the phones will display the name and room number. The extension number and room number are not always the same number.
n	If this field is set to **n**, the phones will display the name and extension number.

Automatic Selection of DID Numbers

This field assigns a 2- to 5-digit number to a guest’s phone number that is not associated with the room number.

Valid entries	Usage
y/n	Enter **y** to use the Automatic Selection of DID Numbers for Guest Rooms feature.

Custom Selection of VIP DID Numbers

This field allows you to select the DID number assigned to a room when a guest checks in. This field can be accessed only if the Automatic Selection of DID Numbers field is **y**.

Valid entries	Usage
y/n	Enter **y** to allow you to select the DID number assigned to a room when a guest checks in.

Number of Digits from PMS

This indicates the number of digits being sent from the PMS to the server to identify room numbers.

NOTE:
If the **Number of Digits from PMS** field is blank and the **PMS Sends Prefix** field is set to **n**, the server will not support an extension that starts with 0.

Valid entries	Usage
1 to **4**	When using normal mode, digits **1** through **4** are valid.
1 to **5**	When using transparent or ASCII mode, digits **1** through **5** are valid.
blank	If using mixed numbering in the server, leave this field blank.

PMS Sends Prefix

This indicates if the PMS sends a prefix digit to the server as part of the room numbering plan.

NOTE:
If the **PMS Sends Prefix** field is set to **n** and the **Number of Digits from PMS** field is blank, the server will not support an extension that starts with 0.

Valid entries	Usage
y/**n**	

Number of Digits in PMS Coverage Path

This indicates whether the coverage paths are **3** or **4** digits long. There can be up to 9999 coverage paths.

Valid entries	Usage
3 to **4**	

Digit to Insert/Delete

Enter the leading digit that may be deleted and inserted back as described in the following text. The current PMS message set uses the extension number as the room identifier. In many customer configurations, the leading digit of the extension number is dropped to screen the room number. In order to accommodate PMS devices that are based on room number and not extension, this leading digit may be deleted on messages from the switch to the PMS, and then inserted back on messages from the PMS.

NOTE:
The PMS interface supports 3-, 4-, or 5-digit extensions, but prefixed extensions do not send the entire number across the interface. Only the assigned extension number is sent. Therefore, you should not use prefixed extensions for numbers that are also going to use the Digit to Insert/Delete function.

Valid entries	Usage
0 to **9**	

Field descriptions for page 3

```
                                                          Page 3 of 3
                  HOSPITALITY

ROOM STATES:

                  Definition for Rooms in State 1: Rooms in State 1
                  Definition for Rooms in State 2: Rooms in State 2
                  Definition for Rooms in State 3: Rooms in State 3
                  Definition for Rooms in State 4: Rooms in State 4
                  Definition for Rooms in State 5: Rooms in State 5
                  Definition for Rooms in State 6: Rooms in State 6

HOSPITALITY FEATURES

                                         Suite Check-in? n
```

Screen 115. Hospitality screen

Definition for Rooms in State 1 - 6

Enter up to a 30-character definition for each room status. For example, you could define state 1 as 'clean, ready to use' and state 2 as 'occupied, needs cleaning.'

The definitions for room states (Field descriptions for page 3), are for Attendant Room Status only. If you are not using Attendant Room Status, you do not need to complete these fields.

Suite Check-in

This field allows attendants to have the system automatically check-in several related extensions with one check-in command.

Valid entries	Usage
y/n	Enter **y** to use the Suite Check-in feature. Refer to ''Suite Check-in'' for more information.

Hunt Group

Allows calls to be answered by users (agents) at a predefined group of telephones or devices.

This screen creates a hunt group that is identified by a hunt group number. Users assigned to a hunt group are identified by their extension number.

This screen can be used to implement a hunt group and its associated features such as Automatic Call Distribution (ACD) and Hunt Group Queuing. Look at the various hunt group screens and choose the screens that can be used to implement your hunt group requirements.

The total number of pages vary depending on your System configuration. Refer to the *Avaya MultiVantage™ Hardware Solutions Guide* for the maximum number of hunt groups supported by each configuration.

The System checks for the busy or idle status of extension numbers in the hunt group when answering calls. A Uniform Call Distribution (UCD) type hunt group selects the “most idle” extension in the group when answering a new call. A Direct Department Calling (DDC) type hunt group selects the first available extension (in the administered sequence) when answering a new call. Expert Agent Distribution (EAD), used only with Expert Agent Selection (EAS), selects the “most idle” agent or the “least occupied” agent with the highest skill level for the call’s skill.

NOTE:
Vector controlled splits/skills can be called directly via the split/skill extension (instead of calling a VDN mapped to a vector that will terminate the call to a vector controlled split/skill); however, the calls will not receive any announcements, be forwarded, redirect to coverage, or intraflow/interflow to another hunt group.

Field descriptions for page 1

```
change hunt-group 4                                            Page 1 of X
                                  HUNT GROUP

           Group Number: 4__                                       ACD? _
             Group Name: ____________________________            Queue? _
        Group Extension: ________                               Vector? _
             Group Type: _______                         Coverage Path: ____
                     TN: ___                 Night Service Destination: _____
                    COR: _                               MM Early Answer? _
      Security Code: ____
   ISDN Caller Disp: ________

           Queue Length: ___
Calls Warning Threshold: ___   Port: x____  Extension: ____
 Time Warning Threshold: ___   Port: x____  Extension: ____
```

Screen 116. Hunt Group screen

```
change hunt-group x                                           Page 1 of X
                                  HUNT GROUP

           Group Number: ___                                      ACD? n
             Group Name: ____________________________           Queue? n
        Group Extension: ____                                  Vector? n
             Group Type: ________                       Coverage Path: ____
                     TN: ___                Night Service Destination: _____
                    COR: _                              MM Early Answer? _
      Security Code: ____
ISDN Caller Display: ________
```

Screen 117. Hunt Group screen when Queue and Vector are n

The two Extension fields display only when the Calls Warning Port and the Time Warning Port fields are **x**.

```
change hunt-group x                                              Page 1 of X
                                     HUNT GROUP

            Group Number: ___                                          ACD? n
              Group Name: ____________________________               Queue? y
         Group Extension: ____ ____                                 Vector? y
         Group Type: ______
                 TN: ___
                     COR: _                              MM Early Answer?
      Security Code: ____
ISDN Caller Display: ________

           Queue Length: ___
Calls Warning Threshold: ___   Port: x___   Extension: ____
 Time Warning Threshold: ___   Port: x___   Extension: ____
```

Screen 118. Hunt Group screen when Queue and Vector are y

The two Extension fields display only when the Calls Warning Port and the Time Warning Port fields are x.

```
change hunt-group x                                               Page 1 of X
                                     HUNT GROUP

           Group Number: ___                                          ACD? n
             Group Name: ____________________________               Queue? y
        Group Extension: ____ ____                                 Vector? n
             Group Type:                                 Coverage Path: ____
                     TN:                      Night Service Destination: _____
                    COR:                                MM Early Answer?
          Security Code: ____
       ISDN Caller Disp: ________

           Queue Length: ___
Calls Warning Threshold: ___   Port: x___   Extension: ____
 Time Warning Threshold: ___   Port: x___   Extension: ____
```

Screen 119. Hunt Group screen when Queue is y and Vector is n

Group Number

This is a display-only field when the screen is accessed using an administration command such as **add** or **change**.

ACD

Indicates whether Automatic Call distribution is used. This field cannot be set to **y** if, on the System-Parameters Customer-Options screen, the ACD field is **n**.

Valid entries	Usage
y	The hunt group will function as an ACD split/skill. AUDIX hunt groups can function as ACD splits/skills.
n	This feature is not desired, even if, on the System-Parameters Customer-Options screen, the ACD field is **y**. When the hunt group is assigned as an ACD split/skill, the hunt group members serve as ACD agents. The agents in this split/skill must log in to receive ACD split/skill calls. If this hunt group is on a remote switch using the AUDIX in a DCS feature, enter **n**.

Group Name

Enter a character string that uniquely identifies the group (for example, "parts dept," "purchasing," or "sales dept").

Valid entries	Usage
28-character string	

Queue

Specifies a queue for the hunt group.

Valid entries	Usage
y/n	Enter **y** so the hunt group will be served by a queue.

Group Extension

Enter an unused extension number to be assigned to the hunt group. The field may not be blank.

Valid entries	Usage
1 to 7 digits	Unassigned extension

Vector

See example screens for fields that display when this field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that this hunt group will be vector controlled. On the System-Parameters Customer-Option screen, the Vectoring-Basic field must be **y** before **y** can be entered here.

Group Type

The group types available depend on what is administered on your System Parameters Customer Options screen for Automatic Call Distribution (ACD), Expert Agent Selection (EAS) and CentreVu Advocate. The table below shows what group types are available depending on your configuration.

Each option uses a different method to select an extension or agent for a call when two or more extensions or agents are available. The second table shows how calls are handled for each group type.

Table 20. Group Types

	circ	ddc	ucd-mia	ead-mia	ucd-loa	ead-loa	pad
ACD=n	x	x					
ACD, Split, Vector = n/y		x	x				
ACD, Skill, Vector = n/y			x	x			
ACD, Skill, Vector = y Advocate or Elite			x	x	x	x	
ACD, Skill, Vector = y Dynamic Advocate			x	x	x	x	x

Valid entries	Usage
ddc	Enter ddc when the call should be routed to the first extension or ACD agent assigned in the ACD split. Group type ddc is also known as “hot seat” distribution. “ddc” distribution is not available when the group is administered as a skill.
ucd-mia ucd-loa	When ucd-mia or ucd-loa is entered, a call routes to the most-idle agent based on when the agent finished the most recent call (“ucd-mia”), or the least occupied agent based on agent occupancy (“ucd-loa”). Enter ucd-mia or ucd-loa if the hunt group has an AUDIX message. One of these entries is required when supporting the Outbound Call Management feature and when the Controlling Adjunct field is asai.
ead-mia ead-loa	When ead-mia or ead-loa is entered, a call routes to the available agent with the highest skill level for the call. If two or more agents with equal skill levels are available, Avaya MultiVantage routes the call to the most-idle agent based on when the agent finished the most recent call (“ead-mia”), or the least occupied agent based on agent occupancy (“ead-loa”). This allows a call to be distributed to the agent best able to handle the call if multiple agents are available.
circ	Enter circ (circular) when the call should be routed in a “round-robin” order. The order in which you administer the extensions determines the order that calls are directed. The switch keeps track of the last extension in the hunt group to which a call was connected. The next call to the hunt group is offered to the next extension in the circular list independent of how long that extension has been idle. You cannot use circular hunting with automatic call distribution, queues, or vectors.
pad	Enter pad (percent allocation distribution) to select an agent from a group of available agents based on a comparison of the agent’s work time in the skill and the agent’s target allocation for the skill.

Coverage Path

Enter a coverage path number. This assigns a coverage path for the hunt group. The coverage path is assigned using the Coverage Path screen. Does not appear if the Vector field is **y**

Valid entries	Usage
1 to **999**	
t1 to **t999**	Time of day table

TN

Enter the Tenant Partition number.

Valid entries	Usage
1 to **20**	

Night Service Destination

Enter the destination where calls to this split will redirect when the split is in the night service mode. Not all features will work correctly if this is not a local extension. Does not appear if the Vector field is **y**.

Valid entries	Usage
An assigned extension number (can be a VDN extension)	
attd	An attendant group code.

COR

Enter the class of restriction (COR) number that reflects the desired restriction for the hunt group. If this is a hunt group supporting the AUDIX in a DCS feature, the CORs on the Hunt Group screen on each switch must be the same.

Valid entries	Usage
0 to **95**	

MM Early Answer

This field applies for systems using Multimedia Call Handling only.

Valid entries	Usage
y/n	The system begins to answer an H.320 call and establish an audio channel before offering the conversion call to the hunt group. This starts billing for the call when the call is first put into queue.

Security Code

Enter a 4-digit security code (password) used for the Demand Print feature.

Valid entries	Usage
4-digit security code	

ISDN Caller Disp

This field is required if, on the System-Parameters Customer-Options screen, the ISDN-PRI or ISDN-BRI Trunks field is **y**.

Valid entries	Usage
grp-name **mbr-name**	Enter **grp-name** or **mbr-name** to specify whether the hunt group name or member name, respectively, will be sent to the originating user.
blank	If the ISDN-PRI or the ISDN-BRI Trunks field is **n**, this field must be blank.

Queue Length

Appears if the Queue field is **y**. Enter the maximum number of calls that can be in the queue at the same time. This field must have an entry when the Queue field is **y**.

Valid entries	Usage
1-200	For DEFINITY Server R configurations
1-99	For DEFINITY Server SI configurations

Calls Warning Threshold

Appears if the Queue field is **y**. Enter the number of calls that can be queued before the System flashes the queue status (feature buttons assigned on agents phones) and the optional Auxiliary Queue Call Warning Threshold lamp assigned to the split/skill. These lamps are lighted steadily when at least one call is in queue and the threshold has not yet been reached.

Valid entries	Usage
Must be less than or equal to the queue length.	This field must not be left blank if Calls Warning Port is assigned a port number.

(Calls Warning) Port

Appears if the Queue field is **y**. Enter the seven-character port number assigned to connect the optional external Auxiliary Queue Call Warning Threshold lamp that will flash when the number of calls in queue has exceeded the queue warning threshold (assigned in Calls Warning Threshold).

NOTE:
This port is assigned to an Analog Line circuit pack or given an “X” designation if an extension is used.

Valid entries	Usage
01 through **44**	First and second characters are cabinet number
01 through **03**	
01	
A through **E**	Third character is the carrier
01 through **20**	Fourth and fifth characters are the slot number
01 through **16**	Sixth and seventh characters are the circuit number

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

(Calls Warning) Extension

Appears if the Queue field is **y** and when the Calling Warning Port and the Time Warning Port fields are **X**. An extension is needed when an X is placed in Calls Warning Port. This extension can be used by the Terminal Translation Initialization (TTI) feature to assign a port to this extension from the port itself. Once Calls Warning Port is assigned a valid port (either via TTI or the **change hunt-group** command), then the extension is removed and considered unassigned.

Valid entries	Usage
	Enter an unassigned extension. This field may not be blank.

Time Warning Threshold

Appears if the Queue field is **y** and when the Calling Warning Port and the Time Warning Port fields are **X**. Enter the time in seconds that a call can remain in the queue before the System flashes the Queue status lamps (feature buttons assigned members phones) and the Auxiliary Queue Time Warning lamp assigned to this split/skill.

Valid entries	Usage
0 to **999**	An entry of **0** provides a warning whenever a call is queued.

(Time Warning) Port

Appears if the Queue field is **y**. Enter the seven-character port number assigned to the Auxiliary Queue Time Warning lamp that flashes when the time entered in Time Warning Threshold has been reached by a call in queue.

NOTE:
This port is assigned to an Analog Line circuit pack or given an "X" designation if an extension is used.

Valid entries	Usage
01 through **44**	First and second characters are the cabinet number
01 through **03**	
01	
A through **E**	Third character is the carrier number
01 through **20**	Fourth and fifth characters are the slot number
01 through **16**	Sixth and seventh characters are the circuit number

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

(Time Warning) Extension

Appears if the Queue field is **y**. An extension is needed when an X is placed in Time Warning Port. This extension can be used by the Terminal Translation Initialization (TTI) feature to assign a port to this extension from the port itself. Once Time Warning Port is assigned a valid port (either via TTI or the **change hunt-group** command), then the extension is removed and considered unassigned.

Valid entries	Usage
	Enter an unassigned extension. This field may not be blank.

Field description for page 2

Page 2 of the Hunt group screen appears only when the ACD field on page 1 is **y**. If the ACD field is **n**, page 3 becomes page 2 and all subsequent page numbers are decreased by one.

The Timed ACW Interval field appears only if, on the System-Parameters Customer-Option screen, the Timed ACW field on page 3 is **y**.

The following screen shows Field descriptions for page 2 with all fields appearing.

```
change hunt-group x                                                Page 2 of X
                                  HUNT GROUP
                      Skill? _          Acceptable Service Level (sec): ___
                        AAS? _       Expected Call Handling Time (sec): ___
                   Measured: ______                 VuStats Objective: ____
       Supervisor Extension: ____              Timed ACW Interval (sec): ___
      Priority on Intraflow? _                 Service Level Supervisor? _
     Inflow Threshold (sec): ____               Level 1 Threshold (sec): ___
        Controlling Adjunct: ____               Level 2 Threshold (sec): ___
Adjunct Link Extension: ____
     Multiple Call Handling: ___________  Redirect on No Answer (rings): ___
                                                        Redirect to VDN: _____
                        Forced Entry of Stroke Counts or Call Work Codes? _
```

Screen 120. Hunt Group screen

```
change hunt-group x                                                Page 2 of X
                                  HUNT GROUP
                  Skill? _  Acceptable Service Level (sec): ___
                    AAS? _
               Measured: ____
   Supervisor Extension: ____
  Priority on Intraflow? _
 Inflow Threshold (sec): __

    Controlling Adjunct: ____
                             Redirect on No Answer (rings): ___
                                                        Redirect to VDN: _____
                        Forced Entry of Stroke Counts or Call Work Codes? _
```

Screen 121. Hunt Group screen when Queue and Vector are n

```
change hunt-group x                                                Page 2 of X
                                  HUNT GROUP
                  Skill? _ Acceptable Service Level (sec): ___
                    AAS? _
               Measured: ______
   Supervisor Extension: ____


    Controlling Adjunct: ____

                             Redirect on No Answer (rings): ___
                                                        Redirect to VDN: _____
                        Forced Entry of Stroke Counts or Call Work Codes? _
```

Screen 122. Hunt Group screen when Vector is n and Queue is y

```
change hunt-group x                                              Page 2 of X
                                   HUNT GROUP
                     Skill? _            Acceptable Service Level (sec): ___
                  AAS? _
             Measured: internal
 Supervisor Extension: ____


  Controlling Adjunct: ____

                           Redirect on No Answer (rings): ___
                                                    Redirect to VDN: _____
                       Forced Entry of Stroke Counts or Call Work Codes? _
```

Screen 123. Hunt Group screen when Queue and Vector are y

Skill

Only appears if, on the System-Parameters Customer-Options screen, the Expert Agent Selection field is **y**.

If this field is **y**, then the Group Type field must be **ucd** or **ead**.

Valid entries	Usage
y/n	Enter **y** if this hunt group is to be an EAS skill.

Acceptable Service Level (sec)

Enter the number of seconds within which calls to this hunt group should be answered. This allows BCMS and/or VuStats to report a percentage of calls that were answered within the specified time. This entry is also used by the CentreVu® Advocate Service Objective feature.

Valid entries	Usage
0 through 9999 seconds	

AAS

Appears when the ACD field is **y**.

Valid entries	Usage
y/n	Enter **y** if this hunt group is to serve as an Auto-Available Split.

Expected Call Handling Time (sec)

Appears only if, on the System-Parameters Customer-Options screen, either the Vectoring (Advanced Routing) or CentreVu Advocate field is **y**.

Valid entries	Usage
1 to **9999**	Establishes the number of seconds for expected call handling. This value is used to initialize Expected Wait Time and is also used by the CentreVu® Advocate Percent Allocation feature.

Measured

Provides measurement data for the ACD split/skill collected (internal to the switch) for VuStats or BCMS. This measurement data is collected for VuStats and BCMS only if, on the System-Parameters Customer-Options screen, they are **y**.

Valid entries	Usage
internal	If you enter **internal** in this field and on the System-Parameters Customer-Options screen neither the VuStats or BCMS field is **y**, the system displays the following message: `<value> cannot be used; assign either BCMS or VuStats first` Contact your Avaya representative to assist with any changes you want to make on the System-Parameters Customer-Options screen.
external	Provides measurements made by the Call Management System (external to switch).
both	Provides measurements collected both internally and externally.
none	Measurement reports for this hunt group are not required.

Service Objective

Appears when Skill and Centre Vu Advocate are y on the Feature Related System Parameters screen.

Valid entries	Usage
1-9999	Enter the per-skill service objective.

VuStats Objective

Enter a numerical user-defined objective. An objective is a split or skill goal for the call. This could be an agent objective such as a specific number of calls handled or an average talk time. The objective could also be a percent within the service level. The objective appears on the VuStats display and allows agents and supervisors to compare the current performance against the value of the objective for the split or skill.

You can use this value in a customized VuStats display format if, on the VuStats display format screen, the Object Type field is either **agent**, **agent-extension**, or **split**.

This field appears only if, on the System-Parameters Customer-Options screen, the VuStats field is **y** and the Measured field is either **internal** or **both**.

Valid entries	Usage
0-99999	Enter a split or skill objective.

Supervisor Extension

Valid entries	Usage
	Enter the extension number (cannot be a VDN number) of the ACD split/skill supervisor that agents will reach when using the Assist feature

Timed ACW Interval (sec)

When a value is entered in this field, an agent in auto-in work mode who receives an ACD call from this hunt group is placed automatically into After Call Work (ACW) when the call drops. Enter the number of seconds the agent should remain in ACW following the call. When the administered time is over, the agent automatically becomes available. Timed ACW cannot be administered if the hunt group is adjunct controlled, is an AUDIX Message Center, or is an auto-available split. The Timed ACW Interval field appears only if, on page 3 of the System Parameters Customer-Option screen, the Timed ACW field is **y**.

Valid entries	Usage
1-9999	The number of seconds the agent should remain in ACW following the call.

Priority On Intraflow

Does not appear if the Vector field is **y**.

Valid entries	Usage
y/n	Enter **y** for calls intraflowing from this split to a covering split to be given priority over other calls waiting in the covering split queue.

Service Level Supervisor

Appears if, on the System Parameters Customer-Options screen, the CentreVu Advocate field is **y** and, on the Hunt Group screen, the ACD and Skill fields are **y**. For information on CentreVu® Advocate, please contact your Avaya Account Executive or refer to the *CentreVu® Advocate User Guide*.

Valid entries	Usage
y/n	Enter **y** to use Service Level Supervisor for this skill.

Inflow Threshold

Appears only when the Vector field is **n** and the Queue field is **y**. Enter the number of seconds that a call can remain in the queue before no more calls will be accepted by the queue. If 0 is entered, a call is redirected to this split/skill only if there is an available agent.

Level 1 Threshold (sec)

Enter the number of seconds corresponding to the Expected Wait Time (EWT) you want to set for this threshold. For example, if you enter 45 calls whose EWT exceeds 45 seconds will be classified as over threshold 1. This field is used with Service Level Supervisor and only appears if the Service Level Supervisor field is **y**.

Controlling Adjunct

If the controlling adjunct is a CONVERSANT voice system (requires an ASAI link), then enter asai in this field. (On the System-Parameters Customer-Option screen, the ASAI Link Core Capabilities and Computer Telephony Adjunct Links fields must be **y** for CallVisor ASAI capability.)

Valid entries	Usage
none	Indicates that members of the split/skill or hunt group are not controlled by an adjunct processor.
asai	All agent logins are controlled by an associated adjunct and logged-in agents can only use their data terminal keyboards to perform phone functions (for example, change work state).
adjlk	Computer Telephony Adjunct Links

Level 2 Threshold (sec)

Enter the number of seconds corresponding to the Expected Wait Time (EWT) you want to set for this threshold. For example, if you enter 60 calls whose EWT exceeds 60 seconds will be classified as over threshold 2. This field is used with Service Level Supervisor and only appears if the Service Level Supervisor field is **y**.

Dynamic Threshold Adjustment

Appears when Service Level Supervisor on the Hunt Group screen is **y** and CentreVu Dynamic Advocate is **y** on the Feature-Related System Parameters screen.

Valid entries	Usage
y/n	

Dynamic Percentage Adjustment

Appears when Group Type on the Hunt Group screen is **pad** and CentreVu Dynamic Advocate is **y** on the Feature-Related System Parameters screen.

Valid entries	Usage
y/n	

Service Level Target

Appears when Dynamic Percentage Adjustment or Dynamic Threshold Adjustment on the Hunt Group screen is **y** and CentreVu Dynamic Advocate is **y** on the Feature-Related System Parameters screen.

Valid entries	Usage
1-99 (percentage) 1-9999 (time in seconds)	Enter the percentage and time components of the service level target.

Dynamic Queue Position

Appears when Skill is **y** on the Hunt Group screen and CentreVu Dynamic Advocate is **y** on the Feature-Related System Parameters screen.

Valid entries	Usage
y/n	Enter **y** to apply the dynamic queue operation to the calls queued to the skill.

Adjunct Link Extension

Appears when the Controlling Adjunct field is **asai** or **adjlk**. Enter the appropriate ASAI Link extension. This field cannot be blank.

Multiple Call Handling

Appears only if, on the System-Parameters Customer-Options screen, the Multiple Call Handling field is **y** and the ACD field on this screen is **y**. This field defines whether the hunt group can have multiple call handling capabilities, and if so, what type.

Valid entries	Usage
none	Agents who are members of that split/skill can only receive an ACD call from that split/skill when the phone is idle.
on-request	Agents in the Multiple Call Handling split/skill can place a non-ACD or an ACD call on hold and select an available work mode. A queued ACD split/skill or direct agent call then is routed to the agent.
many-forced	An ACD call is delivered automatically to an idle line appearance if the agent is in the Auto-In/Manual-In (MI/AI) work mode and an unrestricted line appearance is available.
one-forced	An ACD call is delivered automatically to an idle line appearance if the agent has no other ACD call on the station, is in the Auto-In/Manual-In (MI/AI) work mode, and an unrestricted line appearance is available.
one-per-skill	An ACD call is delivered automatically to an idle line appearance if the agent has no other ACD call for that skill on the station, is in the Auto-In/Manual-In (MI/AI) work mode, and un unrestricted line appearance is available. Valid in an EAS environment and only when the Skill field is **y**.

Redirect on No Answer (rings)

Enter the maximum number of rings before a call will redirect back to the split/skill, or to the administered VDN.

Valid entries	Usage
1 to **20**	
blank	Deactivates Redirect on No Answer.

Redirect to VDN

To redirect a RONA call to a VDN instead of to the split/skill, enter the extension number of the VDN. The administered VDN must be on-premises and must be administered on the system. The VDN can specify a vector that will in turn route to an off-premises VDN. You cannot enter an extension in this field if the Redirection on No Answer (rings) field is blank. Direct Agent calls go to the agent's coverage path if it is administered. If not, the calls go to a VDN.

Forced Entry of Stroke Counts or Call Work Codes

Appears only when the Controlling Adjunct field is **none**.

Valid entries	Usage
y/n	Enter **y** so either a Stroke Count or Call Work Code must be entered for each call answered by an agent when in the Manual-In mode.

Field description for page 3

```
change hunt-group 1                                                Page x of x
                                HUNT GROUP

                           Message Center: rem-vm
                     Voice Mail Extension: ____
      Calling Party Number to INTUITY AUDIX? n
                            LWC Reception: none
```

Screen 124. Hunt Group screen

Message Center

Enter the type of messaging adjunct for the hunt group. Only one hunt group in the System can be administered as **audix**, one as **qsig-mwi**, one as **fp-mwi**, and one as **rem-vm**.

Valid entries	Usage
rem-vm	DCS feature allowing voice mail to be located on another switch
audix	For AUDIX located on this switch

Valid entries	Usage
qsig-mwi	QSIG network allowing voice mail to be located on another switch
fp-mwi	Public network allowing AUDIX to be located on another switch; administrable only when the ISDN Feature Plus field on the System-Parameters Customer-Options screen is **y**.
none	Indicates the hunt group does not serve as a message hunt group.

Voice Mail Extension

This field only appears if the Message Center field is set to **rem-vm**.

Valid entries	Usage
extension	Enter the UDP extension of the voice mail hunt group on the host switch.

Message Center MSA Name

Enter the name of the Message Center MSA. When it appears, it replaces the Message Center AUDIX Name field. Only appears on DEFINITY Server R for hunt groups when the Message Center field is **msa**.

Message Center AUDIX Name

Enter the name of the Message Center AUDIX. Only appears on DEFINITY Server R for hunt groups when the Message Center field is **audix** or **rem-audix**

Primary

Only appears on DEFINITY Server R for hunt groups when the Message Center field is **audix, rem-audix,** or **msa**.

Valid entries	Usage
y/n	Enter **y** to indicate that the specified AUDIX or Messaging Server is the primary adjunct.

Calling Party Number to INTUITY AUDIX

Only appears when the Message Center field is **audix** or **rem-audix**.

Valid entries	Usage
y/n	Enter **y** to send the calling party number to INTUITY AUDIX.

LWC Reception

Defines the destination for Leave Word Calling (LWC) messages left for the hunt group.

Valid entries	Usage
audix	If LWC is attempted, the messages are stored in AUDIX. The Audix Name field must be filled in too.
msa-spe	If LWC is attempted, the messages are stored in the system processing element (spe). The Messaging Server Name field must be filled in too.
none	

AUDIX Name

Enter the name of the AUDIX machine as it appears on the Node Names screen. Only appears on DEFINITY Server R. Add the AUDIX name to the Node Names screen before entering it in this field. For more information on the Node Names screen, refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software*.

Messaging Server Name

Enter the name of the messaging server machine as it appears on the Node Names screen. Only appears for DEFINITY Server R. Use the **change node-names** command to add the AUDIX name to the Node Names screen before entering it in this field. For more information on the Node Names screen, refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software*.

First Announcement Extension

Appears when the Queue field is **y**. Does not appear if the Vector field is **y**.

Valid entries	Usage
Enter a recorded announcement extension number.	This is the announcement the caller receives after being in the queue for the time interval specified in First Announcement Delay. If the call hasn't been answered after the announcement, the caller hears music (only after the first announcement) if Music-on-Hold is provided, or ringing for as long as it remains in the queue. If this is the forced first announcement, the caller always hears ringback after the announcement; otherwise, the caller hears music (if provided).
blank	Leaving this field blank indicates there will be no announcement.

First Announcement Delay (sec)

Enter the number of seconds that a call remains in queue before the associated first announcement is given the calling party. The call retains its place in the queue while the caller is listening to the recorded announcement. If the call hasn't been answered after the announcement, the caller hears music (for first announcement only) if Music-on-Hold is provided or ringing for as long as the call remains in queue. Appears only if the Queue field is **y** and the Vector field is **n**.

Valid entries	Usage
0 through **99**	When **0** is entered, the first announcement is provided immediately to the caller. This value is set automatically to **0** if there is no queue.
blank	This field must be blank if there is no first announcement.

Second Announcement Extension

Appears only when the ACD and Queue fields both are **y** and the Vector field is **n**.

Valid entries	Usage
Enter the extension number assigned to a recorded announcement.	
blank	Leaving the field blank indicates there is no second announcement.

Second Announcement Delay (sec)

Appears only when the ACD and Queue fields both are **y** and the Vector field is **n**. Enter the time in seconds before the call in the queue receives a second recorded announcement or that the second announcement is repeated.

Valid entries	Usage
1 through **99**	It is recommended that, if this split/skill or hunt group is a coverage point for another split/skill, this delay should not be more than 15 seconds.
blank	Leave blank if there is no second announcement.

Second Announcement Recurring

Appears only when the ACD and Queue fields both are **y** and the Vector field is **n**.

Valid entries	Usage
y	The second announcement can be repeated.
blank	Leave blank if there is no second announcement.

Field descriptions for Hunt group for MWI page

```
change hunt-group x                                                Page 3 of X
                                  HUNT GROUP

                           Message Center: _________
                         Voicemail Number: _________________
Routing Digits (e.g. AAR/ARS Access Code): ____

    Calling Party Number to INTUITY AUDIX? n
                            LWC Reception: none
```

Screen 125. Hunt Group for MWI screen

Message Center

Enter the type of messaging adjunct for the hunt group. Only one hunt group in the System can be administered as **audix**, one as **qsig-mwi**, one as **fp-mwi**, one as **rem-audix**, and up to 6 as **qsig-mwi**.

Valid entries	Usage
rem-audix	DCS feature allowing AUDIX to be located on another switch
audix	For AUDIX located on this switch
qsig-mwi	QSIG network allowing AUDIX to be located on another switch
fp-mwi	Public network allowing AUDIX to be located on another switch; administrable only when the ISDN Feature Plus field on the System-Parameters Customer-Options screen is **y**
none	Indicates the hunt group does not serve as a message hunt group.

Voicemail Number

Appears only if, on the System-Parameters Customer-Options screen, the Basic Call Setup and Basic Supplementary Services fields are **y** and the Message Center field on this screen is **qsig-mwi** or **fp-mwi**. The qsig-mwi selection shows the complete number of the AUDIX hunt group on the Message Center switch for QSIG MWI. The fp-mwi selection shows the public network number of the AUDIX hunt group on the Message Center switch.

Valid entries	Usage
Up to 17 digits	Enter the complete AUDIX dial-up number.

Routing Digits (e.g. AAR/ARS Access Code)

Appears only if the Message Center field is **qsig-mwi** or **fp-mwi**. Shows the AAR (most likely for a Message Center type of qsig-mwi) or ARS (most likely for a Message Center type of fp-mwi) access code which when prepended to the AUDIX Complete Number field defines a route to the Message Center switch hunt group containing the line ports to the AUDIX.

Valid entries	Usage

This field cannot be blank.

Calling Party Number to INTUITY AUDIX

Only appears when the Message Center field is **audix** or **rem-audix**.

Valid entries	Usage
y/n	Enter **y** to send the calling party number to INTUITY AUDIX.

LWC Reception

Defines the destination for Leave Word Calling (LWC) messages left for the hunt group.

Valid entries	Usage
audix	If LWC is attempted, the messages are stored in AUDIX. The Audix Name field must be filled in too.
msa-spe	If LWC is attempted, the messages are stored in the system processing element (spe). The Messaging Server Name field must be filled in too.
none	

Field descriptions for pages 4 through X

```
change hunt-group 1                                                Page   4 of  39
                                  HUNT GROUP
          Group Number: 1       Group Extension: 3001        Group Type: ucd
  Member Range Allowed: 1 - 999         Administered Members (min/max): 1  /9
                                            Total Administered Members: 9
GROUP MEMBER ASSIGNMENTS
      Ext      Name (24 characters)            Ext       Name (24 characters)

 1 :________                                14 :_______
 2 :________                                15 :_______
 3 :8344003_ Audix VP #3                    16 :_______
 4 :8344004_ Audix VP #4                    17 :_______
 5 :8344005_ Audix VP #5                    18 :_______
 6 :8344006_ Audix VP #6                    19 :_______
 7 :________                                20 :_______
 8 :________                                21 :_______
 9 :________                                22 :_______
10 :________                                23 :_______
11 :________                                24 :_______
12 :________                                25 :_______
13 :________                                26 :_______

More Members Exist
```

Screen 126. Hunt Group Members screen

NOTE:
Only Pages 1, 2, and 3 appear if the hunt group is skilled.

Group Number

This display-only field shows the number of a hunt group.

Group Extension

This display-only field shows the extension of the hunt group.

Group Type

This display-only field shows the type of the hunt group.

Member Range Allowed

The range of allowed members displays on all member pages. These values vary depending on the particular system and/or configuration.

Administered Members (min/max)

Appears on all member pages. Indicates the minimum and maximum member number administered for this hunt group.

Total Administered Members

Appears on all member pages. Indicates the total number of members administered for this hunt group.

More Members Exist

This display-only field shows there are more members than currently displayed (the current page is not the last page).

At End of Member List

This display-only field shows the current page is also the last page.

Ext

A display-only field if the Controlling Adjunct field is **asai**. Controlled Agent extensions must be entered on the Adjunct Controlled Agent Table screen. The extension cannot be a VDN. The data module cannot be a member of an ACD split/skill.

Valid entries	Usage
Enter the extension number associated with a member in the hunt group or with an associated data module.	If the Controlling Adjunct field is **none**

Name

This display-only field shows the name assigned to the above extension number when it is administered in the System.

Intercom Group

This screen assigns extensions to intercom groups.

```
change intercom-group 1                                          Page 1 of 2
                                 INTERCOM GROUP
                                             Group Number: 1
                                      Length of Dial Code: _

GROUP MEMBER ASSIGNMENTS
              Ext      DC   Name
           1: ________ __
           2: ________ __
           3: ________ __
           4: ________ __
           5: ________ __
           6: ________ __
           7: ________ __
           8: ________ __
           9: ________ __
          10: ________ __
          11: ________ __
          12: ________ __
          13: ________ __
          14: ________ __
          15: ________ __
          16: ________ __
```

Screen 127. Intercom Group

Group Number

This display-only field shows the group's ID number.

Length of Dial Code

This field sets the number of digits that users must dial to access an extension in the group. (On Page 2, this is a display-only field.)

Valid entries	Usage
1	Enter **1** if there are 9 or fewer members.
2	Enter **2** if there are 10 or more members.

Ext

This field assigns an extension to the group.

Valid entries	Usage
an extension number	Enter a physical extension number. You may not enter a VDN in this field.

DC

This field assigns a dial code to an extension. The dial code is the code users must dial to make intercom calls to the corresponding extension.

Valid entries	Usage
1- or 2-digit code	The number of digits entered must exactly match the number assigned in the Length of Dial Code field. For example, if the Length of Dial Code field is set to 2, you must type 1 as 01 in the DC field.

Name

Display-only field. The switch fills in this field with the name from the Station screen.

Related topics

Refer to ''Using phones as intercoms'' on page 473 for instructions.

Refer to ''Intercom'' on page 1604 for a description of the feature.

Inter-Exchange Carrier (IXC) Codes

This screen allows identification of the IXC in the CDR record.

Field descriptions for page 1

```
change ixc-codes                                                Page 1 of 2
                      INTER-EXCHANGE CARRIER CODES
       IXC Codes Assignments (Enter up to 15)
       CDR   IXC                               CDR   IXC
       IXC   Access                            IXC   Access
       Code  Number     IXC Name               Code  Number     IXC Name
       1:    _______    _______________          9:  _______    _______________
       2:    _______    _______________         10:  _______    _______________
       3:    _______    _______________         11:  _______    _______________
       4:    _______    _______________         12:  _______    _______________
       5:    _______    _______________         13:  _______    _______________
       6:    _______    _______________         14:  _______    _______________
       7:    _______    _______________         15:  _______    _______________
       8:    _______    _______________
```

Screen 128. Inter-Exchange Carrier Codes screen

IXC Access Number

Valid entries	Usage
2 to 11 digits, **0** through **9** and *****	Enter the digits dialed or inserted by AAR/ARS into the outpulsed digit string to access the interexchange carrier. No duplicate access numbers are allowed in the table.

IXC Name

Valid entries	Usage
0 to 15 characters	Description to identify the IXC

Field descriptions for page 2

```
                                                    Page 2 of 2
                       IXC Prefix             IXC Code Format
                  1.   ___                    ____
                  2.   ___                    ____
                  3.   ___                    ____
                  4.   ___                    ____
                  5.   ___                    ____
```

Screen 129. Inter-Exchange Carrier Codes screen

IXC Prefix

Valid entries	Usage
1 to 3 digit prefix	
*****	
101	For line 1
10	For line 2

IXC Code Format

Valid entries	Usage
1 to 4 digit code format	
*****	
x	
X	
xxxx	For line 1
xxx	For line 2

Intra-Switch CDR

This screen administers extensions for which Intra-Switch CDR is to be enabled.

NOTE:
Attendants are not allowed to be optioned for the Intra-Switch CDR feature.

If your system can record more than 100 stations, the system only displays two pages of extensions (112 per page) at one time. When you enter the add command to add extensions, the system automatically begins after the last administered extensions. If you enter the change command, the system display begins with the first extension. If you enter the change command with an extension number, the system begins the display with that extension.

When you enter the command list intra-switch-cdr <extension> count x, the system lists “x” switch extensions administered for Intra-Switch CDR beginning with the extension specified by <extension>. For example, if you enter “list intra-switch-cdr 81000 count 500,” the system displays extension 81000 (if it is administered for Intra-Switch CDR) and the next 500 extensions that are administered for Intra-Switch CDR. The display command functions similarly to the change command.

Capacities

The Intra-Switch CDR extension capacities vary from switch to switch. See the Avaya MultiVantage™ Hardware Solutions Guide.

Field descriptions for page 1

```
change intra-switch cdr                                              Page 1 of 2
                          INTRA-SWITCH CDR

Assigned Members:   5     of  1000  administered
 1: 3640000_ 19:________ 37:________ 55:________ 73:________ 91:_______
 2: 3640001_ 20:________ 38:________ 56:________ 74:________ 92:_______
 3: 3641000_ 21:________ 39:________ 57:________ 75:________ 93:_______
 4: 3642001_ 22:________ 40:________ 58:________ 76:________ 94:_______
 5: 3643003_ 23:________ 41:________ 59:________ 77:________ 95:_______
 6: ________ 24:________ 42:________ 60:________ 78:________ 96:_______
 7: ________ 25:________ 43:________ 61:________ 79:________ 97:_______
 8: ________ 26:________ 44:________ 62:________ 80:________ 98:_______
 9:_________ 27:________ 45:________ 63:________ 81:________ 99:_______
10:_________ 28:________ 46:________ 64:________ 82:_______ 100._______
11:_________ 29:________ 47:________ 65:________ 83:_______ 101:_______
12:_________ 30:________ 48:________ 66:________ 84:_______ 102:_______
13:_________ 31:________ 49:________ 67:________ 85:_______ 103:_______
14:_________ 32:________ 50:________ 68:________ 86:_______ 104:_______
15:_________ 33:________ 51:________ 69:________ 87:_______ 105:_______
16:_________ 34:________ 52:________ 70:________ 88:_______ 106:_______
17:_________ 35:________ 53:________ 71:________ 89:_______ 107:_______
18:_________ 36:________ 54:________ 72:________ 90:_______ 108:_______
```

Screen 130. Intra-Switch CDR screen

Assigned Members

Displays the number of extensions currently administered for Intra-switch CDR.

1–x

Valid entries	Usage
Any valid extension	Enter the local extensions you want to track with Intra-Switch CDR. The number of extensions you can track may vary from one system to the next.

IP Codec Set

The IP Codec Set screen allows you to specify the type of codec used for voice encoding and companding (compression/decompression). The main difference between codecs is in the compression algorithm used; some codecs compress the voice data more than others. A greater degree of compression results in lower bandwidth requirements on the network, but may also introduce transmission delays and lower voice quality.

The default codec is set for G711. The G711 provides the highest voice quality because it does the least amount of compression, but it uses the most bandwidth. The G711 default setting can be changed to one of four other codecs if the G711 does not meet your desired voice-quality/bandwidth trade-off specification. Also, if the far-end switch is a not an Avaya MultiVantage, you may need to change the codec to match one that is supported by that switch.

The order in which the codecs are listed on this screen is the order of preference of usage. A trunk call between two DEFINITY switches will be set up to use the first common codec listed.

NOTE:
The codec ordering *must be the same* on DEFINITY switches at both ends of an H.323 trunk connection. The *set* of codecs listed need not be the same, but the *order* of the listed codecs must be the same.

This screen allows you to define the allowed codecs and packet sizes used by each IP network region. You can also enable silence suppression on a per-codec basis. This screen will dynamically display the packet size in milliseconds for each codec in the set, based on the number of frames you administer per packet.

```
change ip-codec-set 1                                               Page 1 of 1

                              IP Codec Set

      Codec Set: 1

     Audio        Silence        Frames    Packet
     Codec        Suppression    Per Pkt   Size (ms)
1: G.711MU_______    y              3         30
2:_______________    _              _
3:_______________    _              _
4:_______________    _              _
5:_______________    _              _
```

Screen 131. IP Codec Set screen

Codec Set

This number identifies the IP codec set.

Valid entries	Usage
1-7	Identifies the IP codec set.

Audio Codec

Specifies the audio codec used for this codec set.

Valid entries	Usage
G.711a (a-law) G.711MU (mu-law) G723.1-6.3 G.723.1-5.3 G.729 G.729B	Enter the codec used for this codec set.

Silence Suppression

Enables RTP-level silence suppression on the audio stream.

Valid entries	Usage
y/n	Enter **y** to enable RTP-level silence suppression on the audio stream.

Frames Per Pkt

Specify the number of frames per packet up to 60 milliseconds.

Valid entries	Usage
1-6	Specify the number of frames per packet up to 60 ms. Note: the frame size for G.711 and G.729 codecs is 1 (10ms), and for G.723.1 is 3 (30ms).

Packet Size (ms)

Displays the packet size in milliseconds.

Valid entries	Usage
10-60	Displays the packet size in milliseconds. Note: the frame size for G.711 and G.729 codecs is 1 (10ms), and for G.723.1 is 3 (30ms).

IP Interfaces

Use this screen to assign a network region to each IP interface device. Use one line for each C-LAN and each IP interface circuit pack.

```
change ip-interfaces                                                Page 1 of 4

                                IP INTERFACES
Inter-region IP connectivity allowed?

Enable                                                                     Net
Eth Pt  Type    Slot    Code Sfx Node Name        Subnet Mask    Gateway Address Rgn
   n   ______ _____          _______________ 255.255.255.0  ___.___.___.___ ___
   n   ______ _____          _______________ 255.255.255.0  ___.___.___.___ ___
   n   ______ _____          _______________ 255.255.255.0  ___.___.___.___ ___
   n   ______ _____          _______________ 255.255.255.0  ___.___.___.___ ___
   n   ______ _____          _______________ 255.255.255.0  ___.___.___.___ ___
   n   ______ _____          _______________ 255.255.255.0  ___.___.___.___ ___
   n   ______ _____          _______________ 255.255.255.0  ___.___.___.___ ___
   n   ______ _____          _______________ 255.255.255.0  ___.___.___.___ ___
   n   ______ _____          _______________ 255.255.255.0  ___.___.___.___ ___
   n   ______ _____          _______________ 255.255.255.0  ___.___.___.___ ___
   n   ______ _____          _______________ 255.255.255.0  ___.___.___.___ ___
```

Screen 132. IP Interfaces screen

Inter-region IP connectivity allowed

Allows IP connections between regions.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints (phones and trunks) to use media processor resources administered in regions that are different from the endpoints' regions.

Enabled Eth Pt

Allows use of the ethernet port.

Valid entries	Usage
y/n	Enter **y** to use the ethernet port. Enter **n** before you make changes on this screen.

Type

Identify the type of IP interface.

Valid entries	Usage
c-lan or **medpro**	Enter the type of IP interface

Slot

Describes the physical port location of the interface on your system.

Valid entries	Usage
Cabinet, carrier and slot location	Enter the slot location of the circuit pack

Code/Sfx

Display only field identifies the interface.

Valid display	Usage
TN799, **TN802**, **TN2302**	Displays the circuit pack code number.

Node Name

The node name must be administered on the Node Names screen.

Valid entries	Usage
Character string	Enter the name of the node.

Subnet Mask

The subnet mask is a 32-bit binary number that divides the network ID and the host ID in an IP address.

Valid entries	Usage
digit string	Identifies the subnet mask associated with the IP address for this IP interface.

Gateway Address

Valid entries	Usage
digit string	Enter the address of a network node that serves as the default gateway for the IP interface. Entering a value in this field poses a potential security risk. For more information on using a default gateway, see *Administration for Network Connectivity for Avaya MultiVantage™ Software*.

Net Rgn

Identify the network region for the interface on this row.

Valid entries	Usage
1-44	Enter the network region number for this interface.

IP Network Region

Field descriptions for page 1

```
change ip-network-region 3                                          Page 1 of 2

                                       IP Network Region
      Region: 10
        Name: North
    Location: 3

             AUDIO PARAMETERS                AUDIO RTCP MONITOR SERVER PARAMETERS
                   Codec Set: 4                           RTCP Reporting Enabled? y
          UDP Port Range Min: 2048_                 Use Default Server Parameters? n
          UDP Port Range Max: 3028                  Server IP Address: 999.999.999.999
          Direct IP-IP Audio? y                                      Server Port: 5005
        IP Audio Hairpinning? y                         RTCP Report Period(secs): 5

DIFFSERV/TOS PARAMETERS                     AUDIO RESOURCE RESERVATION PARAMETERS
      Call Control PHB Value: 34_                                      RSVP Enabled? y
             Audio PHB Value: 46                             RSVP Refresh Rate(secs): 15
                                                  Retry upon RSVP Failure Enabled? y
          802.1P/Q PARAMETERS                        RSVP Profile: guaranteed-service
            802.1p/Q Enabled? y                   RSVP unreserved (BBE) PHB Value: 43
Call Control 802.1p Priority: 7
       Audio 802.1p Priority: 6
                 802.1Q VLAN: 0
```

IP Network Region screen

Region

Displays the number of the region being administered.

Valid entries	Usage
1-44	Numeric identifier for the region.

Name

Description of the region.

Valid entries	Usage
Up to 20 characters	Describes the region.

Location

Specifies the location by IP network region allowing correct date and time information and trunk routing based on IP network region.

Valid entries	Usage
1-44	Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
1-64	(For Avaya S8300 Media Server with an Avaya G700 Media Gateway, Avaya S8700 Media Server for Multi-Connect configuration, and Avaya S8700 Media Server for IP Connect configuration only.) Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and softphones.
blank	The location is obtained from the cabinet containing the CLAN that the endpoint registered through or the media gateway containing the Internal Call Controller or Local Spare Processor on an Avaya S8300 Media Server with an Avaya G700 Media Gateway that the endpoint registered through. This applies to IP telephones and softphones. Traditional cabinets, Remote Offices, and the Avaya S8300 Media Server with an Avaya G700 Media Gateways all have their locations administered on their corresponding screens.

Audio Parameters

Codec Set

Specifies the codec assigned to the region.

Valid entries	Usage
1-7	Enter the number for the codec set for the region.

UPD Port Range Min

Specifies the minimum range of the UDP port number used for audio transport.

Valid entries	Usage
2-65534	Enter the lowest UDP port number to be used for audio transport.

UPD Port Range Max

Specifies the maximum range of the UDP port number used for audio transport.

Valid entries	Usage
3-65535	Enter the highest UDP port number to be used for audio transport.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter to **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack on the switch.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack on the switch in IP format, without going through the DEFINITY TDM bus.

Audio RTCP Monitor Server Parameters

RTCP Reporting Enabled

Valid entries	Usage
y/n	Specifies whether you want to enable RTCP. If this field is set to **y**, then the RTCP Monitor Server Parameters fields appear.

Use Default Server Parameters

This field only appears when the RTCP Enabled field is set to **y**.

Valid entries	Usage
y/n	Enter **y** to use the default RTCP Monitor server parameters as defined on the IP Options System Parameters screen. If you enter **n**, then you need to complete the Server IP Address, Server Port, and RTCP Report Period fields that appear.

Server IP Address

This field only appears when the Use Default Server Parameters field is set to **n** and the and the RTCP Enabled field is set to **y**

Valid entries	Usage
0-255 in nnn.nnn.nnn.nnn format	Enter the IP address for the RTCP Monitor server.

Server Port

This field only appears when the Use Default Server Parameters field is set to n and the and the RTCP Enabled field is set to **y**

Valid entries	Usage
1-65535	Enter the port for the RTCP Monitor server.

RTCP Report Period (secs)

This field only appears when the Use Default Server Parameters field is set to n and the and the RTCP Enabled field is set to **y**.

Valid entries	Usage
5-30	Enter the report period for the RTCP Monitor server in seconds.

DiffServ/TOS Parameters

Call Control PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the IP TOS field to support the DiffServ codepoint.

Valid entries	Usage
0-63	Enter the decimal equivalent of the Call Control PHB value.

Audio PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Use the IP TOS field to support the Audio PHBcodepoint.

Valid entries	Usage
0-63	Enter the decimal equivalent of the DiffServ Audio PHB value.

BBE PHB Value

This field contains the Better than Best Effort (BBE) PHB value.

Valid entries	Usage
0-63	Enter the decimal equivalent of the DiffServ BBE PHB value.

802.IP/Q Parameters

802.1p/Q Enabled

When this field is **y**:

- The `802.1Q VLAN ID` field defaults to zero (0).
- The DSCP value is taken from the Network Region form.
- The Ethernet frame format changes. If corresponding changes are not made to the attached Ethernet switch, service to the C-LAN is interrupted. Ethernet frames will only be tagged with VLAN and user-priority values if this field is **y**.

For all H.323 call signaling

- Ethernet frames are marked with the user priority configured in the `Call Control 802.1p Priority` field.
- IP packets are marked with the VLAN tag configured in the `802.1Q VLAN` field.

NOTE:
802.1Q is not appropriate for a hub-based network.

Valid entries	Usage
y/n	Enter **y** for 802.1p MAC-layer prioritization and 802.1Q Virtual LAN specification for this region.

Call Control 802.1p Priority

Appears only if the 802.1p/Q Enabled field is **y**.

Valid entries	Usage
0-7	Specifies the 802.1p priority value.

Audio 802.1p Priority

Appears only if the 802.1p/Q Enabled field is **y**.

Valid entries	Usage
0-7	Specifies the Audio 802.1p priority value.

802.1Q VLAN

Appears only if the 802.1p/Q Enabled field is **y**.

Valid entries	Usage
0-4095	Specifies the 802.1Q virtual LAN value.

Audio Resource Reservation Parameters

RSVP Enabled

Controls the appearance of the other fields in this section.

Valid entries	Usage
y/n	Specifies whether or not you want to enable RSVP.

RSVP Refresh Rate (secs)

This field only appears if the RSVP Enabled field is set to **y**.

Valid entries	Usage
1-99	Enter the RSVP refresh rate in seconds.

Retry upon RSVP Failure Enabled

This field only appears if the RSVP Enabled field is set to **y**.

Valid entries	Usage
y/n	Specifies whether you to enable retries when RSVP fails.

RSVP Profile

This field only appears if the RSVP Enabled field is set to **y**. You set this field to what you have configured on your network.

Valid entries	Usage
guaranteed-service	
controlled-load	

RSVP unreserved (BBE) PHB Value

This field only appears if the RSVP Enabled field is set to **y**.

Page 2

```
change ip-network-region 10                                      Page 2 of 2

                       Inter Network Region Connection Management

Region                                (Group of 32)
       1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2
001-032                  4
033-064
065-080
```

Screen 133. Inter Network Region Connection Management screen

Region (001-500)

This screen does not appear unless the IP Stations, the H.323 Trunks, and the Remote Office fields on the System-Parameters Customer-Options screen are enabled.

Valid entries	Usage
blank	Indicates there is no connectivity between these network regions.
1-7	Indicates the preferred codec set that you should use between these network regions.

IP Node Names

```
change node-names ip                                              Page 1 of X

                         NODE NAMES

Name             IP Address    Name              IP Address
 1. ______________     ___.___.___.___  17. ______________     ___.___.___.___
 2. ______________     ___.___.___.___  18. ______________     ___.___.___.___
 3. ______________     ___.___.___.___  19. ______________     ___.___.___.___
 4. ______________     ___.___.___.___  20. ______________     ___.___.___.___
 5. ______________     ___.___.___.___  21. ______________     ___.___.___.___
 6. ______________     ___.___.___.___  22. ______________     ___.___.___.___
 7. ______________     ___.___.___.___  23. ______________     ___.___.___.___
 8. ______________     ___.___.___.___  24. ______________     ___.___.___.___
 9. ______________     ___.___.___.___  25. ______________     ___.___.___.___
10. ______________     ___.___.___.___  26. ______________     ___.___.___.___
11. ______________     ___.___.___.___  27. ______________     ___.___.___.___
12. ______________     ___.___.___.___  28. ______________     ___.___.___.___
13. ______________     ___.___.___.___  29. ______________     ___.___.___.___
14. ______________     ___.___.___.___  30. ______________     ___.___.___.___
15. ______________     ___.___.___.___  31. ______________     ___.___.___.___
16. ______________     ___.___.___.___  32. ______________     ___.___.___.___
```

Screen 134. IP Node Names

Name

Identifies the name of the adjunct or switch node.

Valid entries	Usage
1-15 alpha-numeric characters	Used as a label for the associated IP address. The node names must be unique on each switch.

IP Address

The IP address for the node named in the previous field.

Valid entries	Usage
32-bit address (4 decimal numbers, each in the range 1-255)	A unique IP address is assigned to each port on any IP device that is used for a connection. See the *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

IP-Options System Parameters

Field descriptions for page 1

```
display system-parameters ip-options                                      page 1

                              IP-OPTIONS SYSTEM PARAMETERS

         IP MEDIA PACKET PERFORMANCE THRESHOLDS
           Roundtrip Propagation Delay (ms)    High: 800     Low: 400
                         Packet Loss (%)       High: 40      Low: 15
                         Ping Test Interval   (sec): 20
           Number of Pings Per Measurement Interval: 10

         RTCP MONITOR SERVER
                          Default Server IP Address: 123.123.123.123
                                Default Server Port: 5005
                   Default RTCP Report Period(secs): 5

MEDIA GATEWAY ANNOUNCEMENT SERVER PARAMETERS
  Announcement Server IP Address: ___.___.___.___ Login:
                                                Password:
  Announcement Storage Path Name:
```

Screen 135. IP-Options System Parameters screen

RTCP Monitor Server

Default Server IP Address

The default server IP address that can be utilized by the IP Network Region screen for each administered region.

Valid entries	Usage
0-255 in nnn.nnn.nnn.nnn format	A unique IP address is assigned to each port on any IP device that is used for a connection.

Default Server Port

The RTCP monitor is a separate computer that receives RTCP packets from many devices. The MultiVantage software pushes these values to IP phones, IP softphones and VoIP media modules such that they know where to send the data. The IP address is that of the RTCP server. The server port is the TCP/IP port of that RTCP server where the information should be sent.

Valid entries	Usage
1-65535	Enter the port being used as an RTCP monitor.

Default RTCP Report Period(secs)

In conjunction with the IP address and server port, this value tells the IP phones, IP softphones and VoIP media modules how often to send the information (RTCP packets) to the server.

Valid entries	Usage
5-99	

Media Gateway Announcement Server Parameters (available in December)

Announcement Server IP Address

Identifies the IP address of the Announcement Server.

Valid entries	Usage
0-255	A unique IP address is assigned to each port on any IP device that is used for a connection.

Login

Indicates the login to be used by the Media Gateway to access the Announcement Server.

Valid entries	Usage
1-10 characters or blank	Enter a login up to 10 characters.

Password

Indicates the password to be used by the Media Gateway to access the Announcement Server.

Valid entries	Usage
1-10 characters or blank	Enter a password up to 10 characters.

Announcement Storage Path Name

Indicates the path name on the Announcement Server where the announcements are stored.

Valid entries	Usage
Up to 40 characters or blank	Enter the directory path name where announcements are stored.

IP Routing

```
change ip-route 1                                                    Page 1 of 1

                                IP ROUTING

    Route Number:
Destination Node:
    Network Bits:      Subnet Mask:
         Gateway:
           Board:
          Metric:
```

Route Number

Identifies the IP route.

Valid entries	Usage
1-400	Enter the number of the IP route you want to add or change, or enter **n** for the next available number.

Destination Node

The node name of the final destination for this connection.

Valid entries	Usage
The name previously entered on the Node Names screen.	Enter the name of the final destination node of the IP route for this connection.

Network Bits

Valid entries	Usage
0-32	Enter the number of Network Bits that corresponds to the Subnet Mask for the IP Route. If you skip this field and enter a Subnet Mask, the system automatically sets the corresponding Network Bits value. For example, if you set this field to **24**, then the system sets the Subnet Mask field to **255.255.255.0**.

Subnet Mask

Valid entries	Usage
255, 254, 252, 248, 240, 224, 192, 128, 0	Enter the subnet mask associated with this IP route. If you enter a value in the Network Bits field, the system automatically completes this field with the corresponding Subnet Mask. For example, if you set this field to **255.255.255.0**, then the system sets the Network Bits field to **24**.

There is one-to-one mapping between the `Network Bits` and the `Subnet Mask` fields; entering a value in one field uniquely determines the other field. A list of Subnet Mask addresses and their corresponding Network Bits are shown in Table 21).

Table 21. Network Bits and Subnet Mask fields

Network Bits	Subnet Mask	Number of Hosts	Network Type
0	0.0.0.0	4,294,967,294	/ 0
1	128.0.0.0	2,147,483,646	/ 1
2	192.0.0.0	1,073,741,822	/ 2
3	224.0.0.0	536,870,910	/ 3
4	240.0.0.0	268,435,454	/ 4
5	248.0.0.0	134,217,726	/ 5
6	252.0.0.0	67,108,862	/ 6
7	254.0.0.0	33,554,430	/ 7
8	255.0.0.0	16,777,214	/ 8
9	255.128.0.0	8,388,606	/ 9
10	255.192.0.0	4,194,302	/ 10
11	255.224.0.0	2,097,150	/ 11
12	255.240.0.0	1,048,574	/ 12
13	255.248.0.0	524,286	/ 13
14	255.252.0.0	262,142	/ 14
15	255.254.0.0	131,070	/ 15

Continued on next page

Table 21. Network Bits and Subnet Mask fields (Continued)

Network Bits	Subnet Mask	Number of Hosts	Network Type
16	255.255.0.0	65,534	/ 16
17	255.255.128.0	32,766	/ 17
18	255.255.192.0	16,382	/ 18
19	255.255.224.0	8,190	/ 19
20	255.255.240.0	4,094	/ 20
21	255.255.248.0	2,046	/ 21
22	255.255.252.0	1,022	/ 22
23	255.255.254.0	510	/ 23
24	255.255.255.0	254	/ 24
25	255.255.255.128	126	/ 25
26	255.255.255.192	62	/ 26
27	255.255.255.224	30	/ 27
28	255.255.255.240	14	/ 28
29	255.255.255.248	6	/ 29
30	255.255.255.252	2	/ 30
31	255.255.255.254	1	/31
	255.255.255.255	0	/32

NOTE:
For the Network Bits and Subnet Mask fields, if you put a value into either field and then press ENTER or TAB to move the cursor to another field, the other field gets populated automatically with a value corresponding to the one you just entered.

Network Bits

This field is a 32-bit binary number that divides the network ID and the host ID in an IP address.

Gateway

The node name of the first intermediate node.

Valid entries	Usage
A name previously entered on the Node Names screen and is either a port on the CLAN circuit pack or is identified as a Destination Node on another IP route.	If there are one or more intermediate nodes, the first intermediate node is the Gateway. If there are no intermediate nodes between the local and remote CLAN ports for this connection, the Gateway is the local CLAN port.

Board

Valid entries	Usage
1 to **44**	cabinet
A to **E**	carrier
0 to **20**	slot
1 to **44**	gateway
V1 to **V9**	module

Metric

Valid entries	Usage
0 or **1**	Enter **1** on a switch that has more than one CLAN circuit pack installed. See *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

IP Services

```
change ip-services                                                Page 1 of  X

                              IP SERVICES
Service      Enabled    Local      Local         Remote        Remote
  Type                  Node       Port          Node          Port
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
___________    _      __________   ____   ___________________ _____
```

IP Services screen

Service Type

Defines the service provided.

Valid entries	Usage
ALARM1, ALARM2	Use this to connect send alarms over a TCP/IP link.
cbc	Enter cbc to reserve the trunk for outgoing use only to enhance Network Call Redirection.
CDR1, CDR2	Use this to connect either the primary or secondary CDR device over a TCP/IP link.

Valid entries	Usage
DLG	(For Avaya MultiVantage on a DEFINITY Server CSI (CSI), Avaya S8300 Media Server, Avaya S8100 Media Server with Avaya CMC1 Media Gateway (Avaya MultiVantage™ S8100 Media Server with CMC1 Media Gateway), and Avaya S8100 Media Server with Avaya G600 Media Gateway (IP600) only.) Use this command for DEFINITY LAN Gateway. For more information, see the *Avaya MultiVantage CallVisor ASAI Technical Reference*. **NOTE:** If DLG is selected, another screen displays. See “DLG Administration” on page 727.
PMS_JOURNAL	Use this to connect the PMS journal printer over a TCP/IP link.
PMS_LOG	Use this to connect the PMS log printer over a TCP/IP link.
SAT	System administration terminal. Not available on Avaya MultiVantage™ S8100 Media Server with CMC1 Media Gateway.
SYS_PRINT	Use this to connect the system printer over a TCP/IP link.

Enabled

Valid entries	Usage
y/n	Enter **y** to enable this IP service. Only applies to SAT services.

Local Node

Specify the node name for the port.

Valid entries	Usage
Node names as defined on the Node Names screen.	If the link is administered for services over the C-LAN circuit pack, enter a node name defined on the Node Name screen. See *Administration for Network Connectivity for Avaya MultiVantage™ Software* for information on how to administer node names.
processor	Processor is only available for Avaya MultiVantage™ S8100 Media Server with CMC1 Media Gateway.

Local Port

Specify the originating port number. When the Service Type field is **DLG**, this field becomes **5678** and is read-only.

Valid entries	Usage
5000 to **9999**	Use 5111-5117 for SAT applications
	Use 5678 for ASAI
0	For client applications, this defaults to zero.

Remote Node

Specify the switch at the far end of the link for SAT. The remote node should not be defined as a link on the IP Interface or Data Module screens. When the Service Type field is **DLG**, this field is cleared and read-only.

Valid entries	Usage
Node name as defined on the Node Names screen	For SAT, use a node name to provide added security.
any	Use any available node.

Remote Port

Specify the port number of the destination. When the Service Type field is **DLG**, this field is cleared and read-only.

Valid entries	Usage
5000 to **64,500**	Use if this service is a client application, such as CDR or PMS. This must match the port administered on the adjunct, PC or terminal server that is at the remote end of this connection.
0	Default for System Management applications.

IP Services screen (Session Layer Timers page)

Use this screen to enable reliable protocol for TCP/IP links, and to establish other session-layer parameters. This screen only appears if you enter CDR1, CDR2, PMS_JOURNAL, or PMS_LOG in the Service Type field on page 1 or 2.

```
change ip-services                                                    Page   3 of   3

                                SESSION LAYER TIMERS
Service   Reliable   Packet Resp    Session Connect    SPDU    Connectivity
 Type     Protocol      Timer        Message Cntr      Cntr       Timer
CDR1         y          3                   1            1           1
```

Screen 136. IP Services screen (Session Layer Timer page)

Service Type

Identifies the service type for which you are establishing parameters.

Valid entries	Usage
CDR1, CDR2	Use this to connect either the primary or secondary CDR device over a TCP/IP link.
PMS_JOURNAL	Use this to connect the PMS journal printer over a TCP/IP link.
PMS_LOG	Use this to connect the PMS log printer over a TCP/IP link.

Reliable Protocol

Indicates whether you want to use reliable protocol over this link.

Valid entries	Usage
y/n	Use reliable protocol if the adjunct on the far end of the link supports it.

Packet Resp Timer

Valid entries	Usage
1–300	Determines the number of seconds to wait from the time a packet is sent until a response (acknowledgement) is received from the far-end, before trying to resend the packet.

Session Connect Message Cntr

Valid entries	Usage
1–5	The Session Connect Message counter indicates the number of times the switch tries to establish a connection with the far-end adjunct.

SPDU Cntr

Valid entries	Usage
1–5	The Session Protocol Data Unit counter indicates the number of times the switch transmits a unit of protocol data before generating an error.

Connectivity Timer

Valid entries	Usage
1–300	Indicates the amount of time (in seconds) that the link can be idle before the switch sends a connectivity message to ensure the link is still up.

ISDN trunk group

This screen assigns an Integrated Services Digital Network (ISDN) trunk group that supports the ISDN and Call-by-Call Service Selection service selection features. The trunk group provides end-to-end digital connectivity and supports a wide range of services including voice and non-voice services, to which users have access by a limited set of CCITT-defined, standard multipurpose interfaces.

The ISDN trunk group may contain ISDN-PRI or ISDN-BRI interfaces. However, it is not possible to use the two types of interfaces in the same trunk groups. The type of interface is chosen when the trunk members are assigned to the trunk group.

When ISDN-PRI interfaces are used on ISDN trunk groups, they may also be used to support the Wideband Switching feature. This is intended to work with the H0 (384 Kbps), H11 (1536 Kbps), H12 (1920 Kbps), and NXDS0 (128 to 1984 Kbps) data services, and to support high-speed video conferencing and data applications.

Administration commands

When an ISDN trunk connects two switches, set the trunk options identically at both ends of the connection, with the exception of the Trunk Hunt fields. When ISDN-PRI interfaces are used, it is acceptable for both ends to have the Trunk Hunt fields administered as **cyclical**, but if one end is administered as **ascend**, the other end must be administered as **descend**. This helps avoid the possibility of glare conditions. When ISDN-BRI is used, the Trunk Hunt field has to be **cyclical**.

Hardware requirements

ISDN-BRI interfaces are supported by the TN2185 Trunk-side BRI circuit pack (for implementing the user side of the BRI interface) and by the TN556B or TN556C ISDN-BRI Line circuit pack or the TN2198 ISDN BRI (U-LT) Line circuit pack (for the network side of the BRI interface). The TN2464 circuit supports T1 and E1 digital facilities.

ISDN-PRI interfaces are supported by the TN767 circuit pack (for assignment of a T1 signaling link and up to 24 ISDN-PRI trunk group members), or the TN464C or later circuit pack (for assignment of a T1 or E1 signaling link and up to 24 or 31 ISDN-PRI trunk group members, respectively). The TN2464 and TN2207 circuit pack can also be used with ISDN-PRI.

- The D-channel for ISDN-PRI interfaces switches through either the TN765 Processor Interface (PI) circuit pack or the TN778 Packet Control (PACCON) circuit pack. The D-channel for ISDN-BRI interfaces only switches through the TN778 Packet Control (PACCON) circuit pack.

  NOTE:
  You cannot use the TN765 circuit pack with ISDN-BRI interfaces.

- For DEFINITY Server R configurations, the D-channel switches through the TN1655 Packet Interface (PKTINT) circuit pack.
- A TN780 or TN2182 Tone Clock circuit pack provides synchronization for the DS1 circuit pack.

  NOTE:
  The TN767 cannot be used to carry the D-channel if either the TN778 (PACCON) or TN1655 (PKTINT) circuit packs are used to switch the D-channel. However, in these circumstances, the TN767 can be used for NFAS interfaces carrying only B-channels.

How to administer ISDN trunk groups

The table below shows the screens used to administer the TN765 Processor Interface (PI) for DEFINITY Server SI configurations.

Screen	Field
System-Parameters Customer-Options	Version
	ISDN-PRI
	QSIG Optional Features
Feature-Related System Parameters	Send Non-ISDN Trunk Group Name as Connected Name
	Display Connected Name/Number for ISDN DCS Calls
DS1 Circuit Pack	All
Signaling Group	All
Synchronization Plan	All
Data Module	All

Continued on next page

Screen	Field
Processor Channel Assignment for DEFINITY Server SI	All
Interface Links	All
Trunk Group (ISDN)	All
ISDN Numbering - Public/Unknown	All
ISDN Numbering - Private	All
Route Pattern	All
Hunt Groups	ISDN Caller Display
Terminating Extension Group	ISDN Caller Display

The table below shows the screens used to administer the TN778 Packet Control (PACCON) for an DEFINITY Server SI configurations.

Screen	Field
Feature-Related System Parameters	Send Non-ISDN Trunk Group Name as Connected Name? Display Connected Name/Number for ISDN DCS Calls?
System-Parameters Customer-Options	Version ISDN-BRI Trunks ISDN-PRI QSIG Optional Features
Maintenance-Related System Parameters	Packet Bus Maint
Synchronization Plan	All
Trunk Group (ISDN)	All
ISDN Numbering - Public/Unknown	All
ISDN Numbering - Private	All

Continued on next page

Screen	Field
ISDN-BRI Circuit Pack Screen (if using ISDN-BRI interfaces) **or**	All
DS1 Circuit Pack Screen (if using ISDN-PRI interfaces)	All
Route Pattern	All
Hunt Groups	ISDN Caller Display
Signaling Group (if using ISDN-PRI interfaces)	All
Terminating Extension Group	ISDN Caller Display

The table below shows the screens used to administer the TN778 Packet Control (PACCON) for DEFINITY Server R configurations.

Screen	Field
Feature-Related System Parameters	Send Non-ISDN Trunk Group Name as Connected Name?
	Display Connected Name/Number for ISDN DCS Calls?
System-Parameters Customer-Options	Version
	ISDN-BRI Trunks
	ISDN-PRI
	QSIG Optional Features
Synchronization Plan	All
Trunk Group (ISDN)	All
ISDN Numbering - Public/Unknown	All
ISDN-BRI Circuit Pack Screen (if using ISDN-BRI interfaces) **or**	All
DS1 Circuit Pack Screen (if using ISDN-PRI interfaces)	All
ISDN Numbering - Private	All

Continued on next page

Screen	Field
Route Pattern	All
Hunt Groups	ISDN Caller Display
Signaling Group (if using ISDN-PRI interfaces)	All
Terminating Extension Group	ISDN Caller Display

Notes for Above Tables

- System-Parameters Customer-Options — Set the ISDN-BRI Trunks or ISDN-PRI fields to **y**. For a TN778 and if using ISDN-PRI interfaces, set the PRI Over PACCON field to **y**. The QSIG Optional Features fields may be enabled to allow appropriate administration for the Supplementary Service Protocol.
- Feature-Related System-Parameters — Set the Send Non-ISDN Trunk Group Name as Connected Name and Display Connected Name/Number for ISDN DCS Calls fields.
- ISDN-BRI Trunk Circuit Pack — This screen is required if using ISDN-BRI trunk interfaces. Assign all fields as required.
- DS1 Circuit Pack — This screen is required if using ISDN-PRI interfaces.
  - — DS1 (T1) Circuit Pack

    Assign all fields as required. For Facility Associated Signaling, up to 23 ports are available for administration as trunk members in an associated ISDN-PRI trunk group. The 24th port is used as a signaling channel. For Non-Facility Associated Signaling, all 24 ports may be used on certain DS1 circuit packs. The D-channel signaling function for these packs must be provided by a designated DS1 pack on its 24th channel.
  - — E1 Circuit Pack

    Assign all fields as required. For Facility Associated Signaling, up to 30 ports are available for administration as trunk members in an associated ISDN-PRI trunk group. Port number 16 is used as a signaling channel.
- Maintenance-Related System-Parameters — Use this screen only for a TN778. Set the Packet Bus Maint field to **y**.

- ISDN Trunk Group — Enter information in all the fields except the trunk group members. When using ISDN-PRI interfaces, enter the members after you establish the signaling links.
- Signaling Group — This screen is required if ISDN-PRI interfaces are used. Complete all fields. This screen identifies groups of ISDN-PRI DS1 interface B-channels for which a given D-channel (or D-channel pair) will carry the associated signaling information (supports the Facility and Non-Facility Associated Signaling feature). Each DS1 board that is required to have a D-channel must be in a different signaling group by itself (unless D-channel backup is needed, in which case a second DS1 is administered as a backup D-channel). You are not required to select a channel for a trunk group, but if you do, you must have already defined the trunk group as type ISDN.

  NOTE:
  The following three screens, Processor Interface Data Module, Communication Interface Links, and Communication Processor Channel Assignment are used only to support the ISDN-PRI interfaces using PI TN765.

- Processor Interface Data Module — Use this screen only for a TN765. Assign up to 8 interface links using 8 Processor Interface Data Module screens for multi-carrier cabinet systems, and up to 4 links for single-carrier cabinet systems. One Processor Interface Data Module screen must be completed for each interface link to be assigned.
- Communication Interface Links — Use this screen only for a TN765. Assign link numbers 01 to 08 for a multi-carrier cabinet system or links 01 to 04 for a single-carrier cabinet system as required. When first administering this screen in Avaya MultiVantage for ISDN, do not administer the Enable field.
- Communication Processor Channel Assignment — Use this screen only for a TN765. Enter assigned link numbers and assign associated channel numbers to each link. Complete all fields of the screen as required. When first administering this screen in Avaya MultiVantage for ISDN, you need to:
  - First, administer the Interface Links screen, except the Enable field.
  - Second, administer the ISDN fields on the Processor Channel screen.
  - Last, go back to the Interface Links screen and administer the Enable field.
- ISDN Numbering - Public/Unknown — Complete all fields. This screen supports the ISDN Call Identification Display.

- ISDN Numbering - Private — Complete all fields. This screen supports the ISDN Call Identification Display.
- Routing Pattern — Complete all fields including the Supplemental ISDN Routing Information fields as required.
- Hunt Group — Complete the ISDN Caller Display field by entering either **grp-name** or **mbr-name** to specify whether the hunt group name or member name, respectively, is sent to the originating user (supports the ISDN Call Identification Display feature).
- Terminating Extension Group — Complete the ISDN Caller Display field by entering either **grp-name** or **mbr-name** to specify whether the group name or member name, respectively, is sent to the originating user (supports the ISDN Call Identification Display feature).
- Synchronization Plan — Assigns primary and secondary external synchronization sources for the ISDN-BRI Trunk or DS1 circuit pack. Complete all screen fields as required.

Design Considerations

ISDN-BRI and ISDN-PRI interfaces cannot be mixed in the same trunk group. Therefore, consider the following:

- The earliest trunk member (the lowest numbered one) administered is considered correct.
- If an offending member is subsequently found (meaning the first member was BRI and a later member was PRI, or vice versa), the cursor positions on the offending member, and the following error message appears:

  `Cannot mix BRI and PRI ports in the same trunk group.`

Field descriptions for page 1

Many of the fields on the following screens are described in ''Trunk Group'' on page 1167. If a field on this screen is unique, it is listed and defined.

```
add trunk-group next                                               Page 1 of x
                                  TRUNK GROUP

 Group Number: xxx                   Group Type: isdn         CDR Reports: y
   Group Name: OUTSIDE_CALL_______________ COR: 1_       TN: 1__      TAC: ____
    Direction: two-way_      Outgoing Display? n        Carrier Medium: PRI/BRI
  Dial Access? n                Busy Threshold: 99_         Night Service: ____
 Queue Length: 0__
 Service Type: ________________      Auth Code? n           TestCall ITC: rest
Usage Alloc:_________________________Far End Test Line No: _______________
 TestCall BCC: 4                          TestCall Service: ______________
 TRUNK PARAMETERS
          Codeset to Send Display: 6     Codeset to Send TCM,Lookahead: 6
         Max Message Size to Send: 260   Charge Advice: none_____________
   Supplementary Service Protocol: a     Digit Handling(in/out):overlap/overlap
        Digit Treatment: insertion                         Digits: 1234
        Trunk Hunt: cyclical          QSIG Value-Added? n
                                                   Digital Loss Group: __
Calling Number - Delete: ___ Insert: _______________ Numbering Format: _______
               Bit Rate: 1200_        Synchronization: async    Duplex: full
  Disconnect Supervision - In? y  Out? n
  Answer Supervision Timeout: 0__
```

Screen 137. ISDN Trunk Group screen

The Calling Number – Delete, Insert, and Numbering Format fields are the administrable fields for the Calling Line Identification Prefix feature. They appear when the Direction field is incoming or two-way.

Calling Number – Delete

Valid entries	Usage
1 to **15** **all** **blank**	Enter the number of digits, if any, to delete from the calling party number for all incoming calls on this trunk group.

Calling Number – Insert

Valid entries	Usage
1 to **15** **blank**	Enter the specific digits, if any, to add to the beginning of the digit string of incoming calls when the calling party is a member of this trunk group.

Calling Number – Numbering Format

This field indicates the TON/NPI encoding applied to CPN information modified by the CLI Prefix feature. This encoding does not apply to calls originating locally. The Numbering Format field on page 2 of this screen applies to calls originated from this switch.

If this field is blank, Avaya MultiVantage passes on the encoding received in the incoming setup message. If the incoming setup message did not contain CPN information and digits are added, the outgoing message will contain these digits. If the Numbering Format field is blank in this case, the value defaults to pub-unk.

If the Numbering Format field on page 2 of this screen is also administered as **unknown**, the trunk group is modified to “unk-unk” encoding of the TON/NPI. Therefore, this field also must contain a value other than **unknown**.

NOTE:
The values for this field map to the Type of Numbering (TON) and Numbering Plan Identifier (NPI) values shown below.

Valid entries	Type of numbering (TON)	Numbering plan identifier (NPI)
blank	incoming TON unmodified	incoming NPI unmodified
natl-pub	national(2)	E.164(1)
intl-pub	international(1)	E.164(1)
locl-pub	local/subscriber(4)	E.164(1)
pub-unk	unknown(0)	E.164(1)
lev0-pvt	local(4)	Private Numbering Plan - PNP(9)
lev1-pvt	Regional Level 1(2)	Private Numbering Plan - PNP(9)
lev2-pvt	Regional Level 2(1)	Private Numbering Plan - PNP(9)
unk-unk	unknown(0)	unknown(0)

Carrier Medium

This field lets you to specify the type of transport medium interface used for the ISDN trunk group. Appears only when the Group Type field is **isdn** and, on the System-Parameters Customer-Options screen, either the Async. Transfer Mode (ATM) Trunking or H.323 field is **y**.

Valid entries	Usage
ATM	The trunk is implemented via the ATM Interface circuit pack.
IP	The trunk is implemented via the LAN/WAN (C-LAN)/MedPro Interface circuit pack as an H.323 trunk group.
PRI/BRI	The trunk is implemented as a standard DS1 or BRI interface.

Charge Advice

Use this field to accumulate and access charge information about a call. You already must have set the CDR Reports field to **y** or **r** (ring-intvl) before changing this field from its default of none. Remember that receiving Advice of Charge during the call (administered as “automatic” or “during-on-request”) affects system performance because of the increased ISDN message activity on the signaling channel, which may reduce the maximum call capacity.

Valid entries	Usage
none	Enter **none** if you do not want the system to collect Advice of Charge information for this trunk group.
automatic	Enter **automatic** only if your public network sends Advice of Charge information automatically.
end-on-request	Enter **end-on-request** if Avaya MultiVantage must request charge information with each call, and you want to receive only the final call charge.
during-on-request	Enter **during-on-request** if Avaya MultiVantage must request charge information with each call, and you want charges to display during and at the end of a call.

Codeset to Send Display

This field defines the codeset for sending the information element for Display. The value depends on the type of switch to which the user is connected.

Valid entries	Usage
0	CCITT
6	Any other than CCITT or System 85 R2V4, 4E11
7	System 85 R2V4, 4E11

Codeset to Send National IEs

This field defines the codeset for sending the information element (IE) for national IEs. National IEs include all IEs previously sent only in code set 6 (such as DCS IE). Now these national IEs, including Traveling Class Marks (TCMs) and Lookahead Interflow (LAI), can be sent in code set 6 or 7. The value depends on the type of switch the user is connected to.

Valid entries	Usage
6	Other types.
7	System 85 R2V4, 4E11, or newer switch types

NOTE:
A Traveling Class Mark (that is, the user's FRL or the user's trunk group FRL) is passed between tandem nodes in an ETN in the setup message only when the Service Type field is **tandem**. It then is used by the distant tandem switch to permit access to facilities consistent with the originating user's privileges.

Digit Handling (in/out)

This field defines whether overlap receiving and overlap sending features are enabled. Set the field to **overlap** when you want overlap receiving or overlap sending. Set to **enbloc** when you do not want these features enabled. The first field value indicates digit receiving and the second value indicates digit sending. There are 4 possible combinations: **enbloc/enbloc**, **enbloc/overlap**, **overlap/enbloc**, and **overlap/overlap**.

Without overlap receiving or sending enabled, the digits on incoming and outgoing calls are sent enbloc. If the Digit Handling field is **overlap/enbloc** or **overlap/overlap**, the following results:

- Incoming Call Handling Treatment table does not appear
- The Digit Treatment and Digits fields appear
- Warning message indicates that all Incoming Call Handling entries are removed when screen is submitted
- When screen is submitted with these values, all Incoming Call Handling entries are removed

Far End Test Line No.

Specifies the number sent to the far-end's ISDN test line extension. When the test trunk long command is issued, this exact number is sent to the far-end to establish a call that tests the integrity of the trunk member under test. The number does not pass through routing or undergo digit manipulation. The digits entered must be what the far-end expects. For example, for an ISDN tandem trunk, the far-end test number should be a 7-digit ETN (Electronic Tandem Network) number. Up to 15 digits may be entered in this field.

Max Message Size to Send

Defines the maximum size of ISDN messages sent by the switch. Currently, the system can receive 260 byte messages. Valid entries are **128**, **244**, **256**, and **260**.

The following table indicates the expected ISDN-PRI message size from several Avaya products.

Products	Message Length (octets) Received
4ESS (4E11)	256
4ESS (4E13)	256
4ESS (4E14)	256
5ESS (5E4)	244
5ESS (5E5)	244
5ESS (5E6)	244
System 75 (all)	260
System 85 (R2V4)	128
System 85 (R2V5)	260
System 85 (R2V6)	260

QSIG Value-Added

Provides QSIG-VALU services. This field appears only if the Value-Added (VALU) field on the System-Parameters Customer-Options screen is **y**. This field can be set to **y** only if the Supplementary Service Protocol field on the System-Parameters Customer-Options screen is **b**. Valid entries are **y** and **n**. Blank is not a valid entry.

Service Type

Indicates the service for which this trunk group is dedicated. The following table provides a listing of predefined entries. In addition to the Services/Features listed in this table, any user-defined Facility Type of 0 (feature) or 1 (service) on the Network Facilities screen is allowed.

Up to 10 (DEFINITY Server SI, DEFINITY Server CSI) or 200 (DEFINITY Server R) ISDN trunk groups can have this field administered as **cbc**.

Valid entries	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message oriented signaling.
i800	International 800 Service — allows a subscriber to receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user’s service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.

Continued on next page

Valid entries	Usage
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is public-ntwrk, Dial Access can be set to ***y***.
sddn	Software Defined Data Network — provides a virtual private line connectivity via the AT&T switched network (4ESS switch). Services include voice, data, and video applications. These services complement the SDN service. Do not use for DCS with Rerouting.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.
tandem	Tandem tie trunks integral to an ETN
tie	Tie trunks — general purpose
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.

Supplementary Service Protocol

Indicates which supplementary service protocol to use for services over this trunk group. Supplementary Service protocols are mutually exclusive.

Valid entries	Usage
a	AT&T
b	ISO/ETSI QSIG Private Network
c	ETSI public network
d	European Computer Manufacturer's Association (ECMA) QSIG private network (supports only Name Identification and Additional Network Feature Transit Counter (ANF-TC))
e	DCS with Rerouting — Do not use the Service Type field entry of **dmi-mos** or **sddn** with this option. — Set the Used for DCS field (on page 2) to **y**.
f	ISDN Feature Plus Public network feature plus signaling.
g	ANSI. Available only if, on the System-Parameters Customer-Options screen, the ISDN-PRI or ISDN-BRI field is **y** or the Used for DCS field is **y**.

Test Call BCC

Indicates the Bearer Capability Code (BCC) used for the ISDN test call.

Valid entries	Usage
0	Voice
1	Mode 1
2	Mode 2 Asynchronous
3	Mode 3 Circuit
4	Mode 0

Testcall ITC

Controls the encoding of the Information Transfer Capability (ITC) codepoint of the bearer capability IE in the SETUP message when generating an ISDN test call. Allowed values are **rest** (restricted) and **unre** (unrestricted).

NOTE:
ISDN Testcall feature has no routing, so a testcall is never blocked due to an incompatible ITC.

Testcall Service

Specifies the call-by-call selection for an ISDN test call. Only appears if the Service Type field is cbc. Valid entries are all of the services listed in ''Service Type'' on page 906, excluding **sddn** or any new Facility Type of 0 (feature), 1 (service), or 3 (outgoing) that is defined by users on the Network Specific Facility Encoding screen.

Trunk Hunt

The switch performs a trunk hunt when searching for available channels within a facility in an ISDN trunk group. Enter **ascend** to enable a linear trunk hunt search from the lowest to highest numbered channels, or **descend** for a linear trunk hunt search from the highest to lowest numbered channels. With both **ascend** and **descend**, all trunks within an ISDN trunk group are selected based on this field and without regard to the order in which trunks are administered within the trunk group.

Enter **cyclical** to enable a circular trunk hunt based on the sequence the trunks were administered within the trunk group. When using ISDN-BRI interfaces, only **cyclical** is allowed.

NOTE:
The cyclical option cannot be set if the trunk group using ISDN-PRI interfaces is to be used for Wideband operations (the Wideband Support field set to **y**).

The search can be administered per ISDN-PRI trunk group, but it infers the direction of search within all ISDN-PRI facilities (or portions of those facilities) administered within the trunk group.

Usage Alloc

Appears when the Service Type field is **cbc**.

Valid entries	Usage
y/n	Enter **y** to allocate service provided by the trunk group. Use **y** to enhance Network Call Redirection. When you enter **y**, the CBC Trunk Group Usage Allocation Plans screen and the CBC Trunk Group Usage Allocation Plan Assignment Schedule appear.

Field descriptions for page 2

```
add trunk-group next                                            Page 2 of x
TRUNK FEATURES
          ACA Assignment? n           Measured: none____
                                                           Maintenance Tests? y
                                Data Restriction? n


    Abandoned Call Search? n
    Suppress # Outpulsing? n

        Charge Conversion: 1
            Decimal Point: none
          Currency Symbol:
              Charge Type: units
                                                   Per Call CPN Blocking Code:
                                                 Per Call CPN Unblocking Code:

              Outgoing ANI:                               Ds1 Echo Cancellation? n
```

Screen 138. ISDN Trunk Group screen

```
add trunk-group next                                            Page 2 of x
TRUNK FEATURES
          ACA Assignment? n           Measured: none____   Wideband Support? n
  Long Holding Time(hours: _      Internal Alert? _        Maintenance Tests? y
 Short Holding Time (sec): _    Data Restriction? n     NCA-TSC Trunk Member: 7
  Short Holding Threshold: __          Send Name: n      Send Calling Number: n
             Used for DCS? n             Hop Dgt: _    Send Connected Number: n
    Suppress # Outpulsing? n    Numbering Format: ______     DCS Signaling: ____
Outgoing Channel ID Encoding: _________
        Charge Conversion: 1____                  UUI IE Treatment: shared
            Decimal Point: none__   Maximum Size of UUI IE Contents: 128
          Currency Symbol: ___
              Charge Type: units__        Send Called/Busy/Connected Number: n
                Send UCID? n                BSR Reply-best DISC Cause Value: 31
  Send Codeset 6/7 LAI IE? y
```

Screen 139. ISDN Trunk Group screen

These fields are unique to the ISDN trunk group screen.

BSR Reply-best DISC Cause Value

Switches that are polled as resources in a Best Service Routing application return data to the polling switch in the ISDN DISC message. Since some cause values do not work over some networks, this field sets the cause value that your switch will return in response to a BSR status poll. If this field is set incorrectly, incoming status poll calls over this trunk group will be dropped before any data is returned to the polling switch. This field only appears if the UUI IE Treatment field is set to **shared**.

Valid entries	Usage
31 (normal-unspecified) **17** (user-busy) **16** (normal-call-clearing)	Enter **31** unless otherwise instructed by Avaya or your network service provider.

CAUTION:

In most cases, this field will be set to the appropriate value during installation. If you need to change it, your network service provider should be able to help you choose an appropriate value. Don't change this field without the assistance of Avaya or your network service provider.

DCS Signaling

Specifies the means used to send the DCS message. This field only appears if the Used for DCS field entry is **y** and the Service Type field is anything except **dmi-mos** or **sddn**. Valid entries are **bx.25** for the traditional DCS feature or **d-chan** for the DCS over ISDN-PRI D-channel feature.

DCS over D-channel is not supported on trunk groups containing ISDN-BRI interfaces.

- Hop Dgt — The Tandem Hop Limitation and QSIG Additional Network Feature Transit Counter (ANF-TC) features provide a counter that reflects the number of switches (that is, the number of hops) that a call has gone through. The counter increments as a call leaves Avaya MultiVantage using tandem facilities. Valid values are **y** and **n**. One or both of the features can be applied to the trunk group depending on the following:
  - If you enter **y** and the Group Type field is **tandem** or the Group Type field is **isdn** and the Service Type field is **tandem**, the Tandem Hop Limitation feature is applied to the trunk group.

— If you enter **y** and you set the Group Type field to **isdn**, set the Service Type field to **access**, **dmi-mos**, **public-ntwrk**, **tandem**, **tie**, or any of the craft-defined services allowed in the field. Set the Supplementary Service Protocol field to **b** or **d**, then the ANF-TC feature is applied to calls on the trunk group.

NOTE:
The above conditions overlap. If the Group Type field is **isdn**, the Service Type field is **tandem**, and the Supplementary Service Protocol field is **b** or **d**, then both the Tandem Hop Limitation and ANF-TC features are applied to calls on the trunk group.

— If both features are applied to calls on the trunk group, ANF-TC takes precedence. In situations where Avaya MultiVantage is an Incoming or Outgoing Gateway, either feature uses the hop count and transit information provided by the other.

Maximum Size of UUI IE Contents

This field appears when the UUI IE Treatment field is **shared**. Enter the maximum number of bytes of user information that the network supports.

NCA-TSC Trunk Member

Identifies the trunk member whose D-channel will be used to route tandem NCA-TSCs or QSIG CISCs. Value range for this field is from 1 to the maximum number of members per trunk group supported on the switch.

Network (Japan) Needs Connect Before Disconnect

Sends an ISDN Connect message just prior to the Disconnect message.

Numbering Format

This field appears if the Send Calling Number field is **y** or **r** or the Send Connected Number field is **y** or **r**. This specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. Valid entries are **public**, **unknown**, **private**, and **unk-pvt**. **Public** indicates that the number plan according to CCITT Recommendation E.164 is used and that the Type of Number is national. **Unknown** indicates the Numbering Plan Indicator is unknown and that the Type of Number is unknown. **Private** indicates the Numbering Plan Indicator is PNP and the Type of Number is determined from the ISDN Private-Numbering screen. An entry of **unk-pvt** also determines the Type of Number from the ISDN Private-Numbering screen, but the Numbering Plan Indicator is unknown.

Outgoing Channel ID Encoding

Appears only if the Group Type field is **isdn** and the Service Type field is anything except **dmi-mos** or **sddn**. Determines whether to encode the Channel ID IE as preferred or exclusive. Blank is not a valid entry. Defaults are determined as follows:

If the Group Type field is **isdn** and the Used for DCS field is **y**, default is **exclusive**.

If the Group Type field is **isdn** and the Used for DCS field is **n**, default is **preferred**.

If the Group Type field is not **isdn** or it is **isdn**, but the Used for DCS field does not appear, default is **preferred**.

Path Replacement Method

Appears when either the ISDN-PRI trunk or the ISDN-BRI trunk fields and the Basic Call Setup and Supplementary Services with Rerouting fields are set to **y** on the System-Parameters Customer-Options screen and when the Supplementary Service Protocol is either **b** or **e** and the Group Type field is **isdn** on the ISDN trunk group screen.

Valid entries	Usage
better-route	Uses the most economical route, for example, the reconfigured call does not use the same trunk group as the original call.
always	Always reconfigures the call regardless of the trunk group used.

Replace Restricted Numbers

Appears when the Group Type field is **isdn**. Indicates whether to replace restricted numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. This field applies to BRI and PRI trunks.

Valid entries	Usage
y/n	Enter **y** for the display to be replaced regardless of the service type of the trunk.

Replace Unavailable Numbers

Appears when the Group Type field is **isdn**. Indicates whether to replace unavailable numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. This field applies to BRI and PRI trunks.

Valid entries	Usage
y/n	Enter **y** for the display to be replaced regardless of the service type of the trunk.

Send Called/Busy/Connected Number

This field appears only if the QSIG Value-Added field on the Trunk Group screen is **y** and the Group Type field is **isdn**. Specifies if the connected party's number is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If set to **y**, the ISDN Numbering - Public/Unknown Format screen is accessed to construct the actual number sent, or the ISDN Numbering-Private screen (based on Numbering Format) is used. If set to **r**, the connected number is sent "presentation restricted."

Send Calling Number

Specifies whether the calling party's number is sent on outgoing or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the ISDN Numbering - Public/Unknown Format screen is accessed to construct the actual number to be sent, or the ISDN Numbering-Private screen (based on the Numbering Format field) is used. If the value is **r**, the calling number is sent "presentation restricted."

NOTE:
The ISDN Numbering - Public/Unknown Format screen can override the Send Calling Number field entry for any administrable block of extensions.

Send Codeset 6/7 LAI IE?

Specifies whether the ISDN trunk should transmit information in Codeset 6/7. If the UUI IE Treatment field is **shared**, then this field should be **n**. Otherwise, the same information will be sent twice and may exceed the message size. Default is **y** for pre-DEFINITY 6.3 compatibility.

Send Connected Number

Appears if the QSIG Value-Added field on the Trunk Group screen is **n**. Specifies if the connected party's number is sent on incoming or tandemed ISDN calls. Valid entries are **y**, **n**, or **r** (restricted). If **y** is entered, the ISDN Numbering - Public/Unknown Format screen is accessed to construct the actual number sent, or the ISDN Numbering-Private screen (based on the Numbering Format field) is used. If the value is **r**, the connected number is sent "presentation restricted."

NOTE:
The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/protocol-version combination on the DS1 screen, you should not administer the Send Connected Number field to **r** (restricted) on the ISDN Trunk Group screen, as this causes display problems.

NOTE:
The ISDN Numbering - Public/Unknown Format screen overrides the Send Connected Number field entry for any administrable block of extensions.

Network Call Redirection

Administrable if, on the System-Parameters Customer-Options screen, the ISDN-PRI or ISDN-BRI Trunks field is **y**, the ISDN Network Call Redirection field is **y**, and on the ISDN Trunk Group screen, the Supplementary Service Protocol field is **c** or **g. W**henever the Supplementary Service Protocol field is changed, this field resets to **none** to prevent an inadvertent incorrect value,

Valid entries	Usage
ANSI-transfer	Indicates standard ANSI encoding functionality. Allowed only if the Supplementary Service Protocol field is **g**.
Nortel-transfer	Indicates non-standard ENCT functionality. Allowed only if the Supplementary Service Protocol field is **g**.
deflect	Indicates network call redirection. Allowed only if the Supplementary Service Protocol field is **c**.
none	No redirection (default).

Send Name

Specifies whether the calling/connected/called/busy party's administered name is sent to the network on outgoing/incoming calls. Valid entries are **y**, **n**, or **r** (restricted). The value **r** indicates that the calling/connected name will be sent by the switch but will be marked "presentation restricted." This value is valid only if the Supplementary Service Protocol field is **a** (national supplementary service), **b** (for called/busy only) or **d** for the QSIG Global Networking Supplementary Service Protocol. When the Supplementary Service Protocol field is **e** (DCS with Rerouting), only values of **y** and **n** are permitted.

NOTE:
If name information is not administered for the calling station or the connected/called/busy station, the system sends the extension number in place of the name.

Send UCID

Specifies whether or not the trunk should transmit Universal Call IDs. The valid entries are **y** and **n**.

Send UUI IE

Specifies whether to block sending UUI information on a per trunk group basis. The valid entries are **y** and **n**.

US NI Delayed Calling Name Update

Administrable if, on the System-Parameters Customer-Options screen, the ISDN-PRI field is **y**, and on the Trunk Group screen, the Carrier Medium field is either **PRI/BRI** or **ATM**, and the Supplementary Service Protocol field is **a**. This field provides display updates to the terminating phone for delayed calling party name provided by the network,

Valid entries	Usage
y	If calling name information is received after the incoming call has been delivered to the terminating phone, there is a display update. **NOTE:** BRI trunks do not support this value.
n	If calling name information is received after the incoming call has been delivered to the called phone, there is no display update to the terminating phone.

UUI IE Treatment

Specifies whether the user Information Element (IE) is shared.

Valid entries	Usage
shared	If the trunk is connected to a DEFINITY 6.3 (or later) switch.
service-provider	If the trunk is connected to a pre-DEFINITY 6.3 switch or service provider functionality is desired.
none	Blocks sending UUI information.

Wideband Support

Specifies whether Wideband Switching is supported by this trunk group. Valid entries are **y** or **n**. For ISDN trunk groups containing ISDN-BRI interfaces, the only valid entry is **n**. Otherwise you can administer this field only if the Wideband Switching field is **y** on the System-Parameters Customer-Options screen. If set to **y**, the Wideband Support Options page appears. All trunk members must be from TN464C (or later) circuit packs.

NOTE:
Wideband trunk calls are treated as a single trunk call when Automatic Circuit Assurance (ACA) measurements are taken. This way, if an ACA referral call is generated (for short or long holding time), the wideband call only triggers a single referral call using the lowest B-channel trunk member associated with the wideband channel.

Field descriptions for the Shared UUI Feature Priorities page

The fields in this page show the priorities for each type of information to be forwarded in the Shared UUI. This page appears only on the ISDN trunk group screen when all of the following conditions are met:

- The UUI IE Treatment field is **shared**.
- The Supplementary Service Protocol field is set to anything except **b**.

```
add trunk-group next                                                    Page y
of x
                         SHARED UUI FEATURE PRIORITIES

                       ASAI: 1

         Universal Call ID: 2

MULTI SITE ROUTING (MSR)

               In-VDN Time: 3
                  VDN Name: 4
          Collected Digits: 5
     Other LAI Information: 6
```

Screen 140. Shared UUI Feature Priorities

Changing the priorities in this screen may affect whether certain information will be sent. For more information about setting priorities, refer to “Information Forwarding” in *Avaya MultiVantage™ Call Center Software Release 11 Guide to ACD Call Centers*. These fields are unique to the ISDN trunk group screen.

ASAI

User information from ASAI. Valid entries are **1** to **6** and blank. If blank, that field’s information is not forwarded.

Universal Call ID

Unique tag to identify each call. Valid entries are **1** to **6** and blank. If blank, that field’s information is not forwarded.

In-VDN TIme

Number of seconds the call has spent in vector processing. Valid entries are **1** to **6** and blank. If blank, that field’s information is not forwarded.

VDN Name

Name of the active VDN (also called LAI DNIS). Valid entries are **1** to **6** and blank. If blank, that field’s information is not forwarded.

Collected Digits

Digits collected from caller (not including dial-ahead digits). Valid entries are **1** to **6** and blank. If blank, that field’s information is not forwarded.

Other LAI Information

Includes the time stamp of when the call entered the current queue, the call's priority level in its current queue, and the type of interflow. Valid entries are **1** to **6** and blank. If blank, that field's information is not forwarded.

Field descriptions for the Incoming Call Handling Treatment page

```
add trunk-group next                                          Page y of x
                        INCOMING CALL HANDLING TREATMENT
     Service/    Called     Called       Del  Insert          Per Call    Night
     Feature      Len       Number                             CPN/BN     Serv
   _____________ ___  _______________ ___  ______________ __________ _______
   _____________ ___  _______________ ___  ______________ __________ _______
   _____________ ___  _______________ ___  ______________ __________ _______
   _____________ ___  _______________ ___  ______________ __________ _______
   _____________ ___  _______________ ___  ______________ __________ _______
   _____________ ___  _______________ ___  ______________ __________ _______
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   _____________ ___  _______________ ___  ______________ __________ _______
   _____________ ___  _______________ ___  ______________ __________ _______
   _____________ ___  _______________ ___  ______________ __________ _______
   _____________ ___  _______________ ___  ______________ __________ _______
   _____________ ___  _______________ ___  ______________ __________ _______
```

Screen 141. Incoming Call Handling Treatment

NOTE:
This page does not appear if the Digit Handling (in/out) field is **overlap** on the "in" side or if the Direction field is **outgoing**.

The Incoming Call Handling Treatment page can (optionally) provide unique call treatment for different incoming calls on any ISDN trunk group. The number of unique treatments that can be specified in this table and the number of pages vary depending on whether the Usage Allocation Enhancements feature is enabled and on the amount of available system memory.

Unique digit manipulation capabilities, CPN/BN requests, and night service destinations are possible for different types of incoming calls. The unique digit manipulation capabilities can be particularly useful to accommodate different dial plans for different services on an ISDN trunk type with a Service Type (field entry) of **cbc** (Call-by-Call). The table can also be used for ISDN trunk groups that are not Call-by-Call. For example, an ISDN group with Service Type set to **mega800** could use the Incoming Call Handling Treatment table to distinguish treatment of calls to different 800 numbers based on the Dialed Number Identification Service (DNIS) number that is incoming to the switch.

Each row in the table consists of seven columns. The first three columns (Service/Feature, Called Len, and Called Number) constitute a key that together select which row or unique treatment should apply for an incoming call on the group. The remaining four columns (Del, Insert, and so on) specify the treatment to be provided for a call that matches the key.

If an incoming call is for a service listed in a row on the table, then that row may specify the treatment for the call, depending on the other two columns of the key. The Called Len field is used to continue the row determination. If the number of digits received with the incoming call matches the number of digits in the Called Len field for calls to the matched service, then this row may apply. If no other row also contains a matching service and called length, then this row does apply. If another row does exist with the same service and number length, then the Called Number field will be used to continue the row determination.

If the leading digits received with the incoming call match the digits specified in the Called Number field, then this row applies to the call. Therefore, with this table, a unique treatment can be given to any incoming call, even if these calls are to the same service or have the same length of digits. The remaining four fields specify the unique treatment for the call once the row has been determined. Together, the Del and Insert fields can be used to manipulate the incoming number that will be used to route the call. The Per Call CPN/BN field can be used to request CPN/BN from AT&T networks for specific calls incoming on the group. The Night Serv field is used to have calls of different types routed to different night destinations when night service is in effect.

The Incoming Call Handling Treatment Table always automatically rearranges to show the precedence order the system uses to pick an entry. Thus, you can easily predict the behavior of the Incoming Call Handling Treatment Table by noting the order in which the entries display. (The entries rearrange after submitting the trunk group screen. A subsequent change trunk-group or display trunk-group command then shows the rearranged order.)

Avaya MultiVantage traverses the table from top to bottom and picks the first entry that matches all the following criteria:

- The Service /Feature, if applicable, matches
- The Called/Length matches
- The Called Number matches

If the administered Called Length or Called Number is blank, that criterion is considered successful.

Incoming Call Handling Treatment Table entries with a predefined service/feature always appear before entries with a user-defined service/feature. To control the order in which certain entries appear, you must use user-defined services/features for those entries. For example, you can redefine the predefined mega800 service/feature as an identical user-defined entry with the name m800.

User-defined entries are always listed in the reverse order compared to the way they appear on the ISDN Network-Facilities screen. See ''Packet Gateway Board'' on page 1018. Thus, given two user-defined services/features ABC and XYZ, you can force XYZ to appear before ABC in an Incoming Call Handling Treatment Table by putting XYZ after ABC on the Network-Facilities screen.

NOTE:
DCS features that use the remote-tgs button (on the remote switch) do not work when the local trunk group deletes or inserts digits on the incoming call. These buttons try to dial a local TAC. Adding or deleting digits defeats this operation and renders the remote feature inoperable. If digit manipulation is needed, use it on the outgoing side, based on the routing pattern. One reason for digit manipulation is insertion of the AAR feature access code (FAC).

These fields are located on the Incoming Call Handling Treatment Table screen.

Service/Feature

Specifies the ISDN Services/Features for an incoming call type. See the Service Type field description for a list of predefined Services/Features that can be received. Or, use Type 0, Type 1, and Type 2 user defined services. The identifier other can be used for any Services/Features not explicitly specified.

Called Len

Specifies the number of digits received for an incoming call. A blank entry is used as a “wild card” entry and, when used, means that any length of digits associated with the specified Service/Feature can match in this field. Valid entries are **0** to **21**, or leave blank.

Called Number

Specifies the leading digits received for an incoming call. A blank entry is used as a “wild card” entry and, when used, means that any number associated with the specified Service/Feature can match in this field. Valid entries are **1** to **16**, or leave blank.

Del

Specifies the number of leading digits to be deleted from the incoming Called Party Number. Calls of a particular type may be administered to be routed to a single destination by deleting all incoming digits and then administering the Insert field with the desired extension. Valid entries are **1** to **21**, **all**, or leave blank.

Insert

Specifies the digits to be prepended to the front of the remaining digits after any (optional) digit deletion has been performed. The resultant number formed from digit deletion/insertion is used to route the call, provided night service is not in effect. Valid entries are up to 16 characters consisting of a combination from the following: **0** to **9**, *****, **#**, or leave blank.

Per Call CPN/BN

Specifies when and how to request Calling Party Number (CPN) or Billing Number (BN) for calls of this type. Leave blank when connected to another switch, or when connected to a public network outside North America. Within North America, leave blank when connected to a public network that does not permit customer equipment to request CPN or BN for individual incoming calls. The AT&T Switched Network offers this service under the titles “CPN/BN to Terminating End on a Per-Call Basis” and “ANI (BN) on Request.” An entry of **none** indicates the switch will not request either CPN or BN for any incoming calls of this type. Valid entries are **cpn-only**, **bn-only**, **bn-pref** (prefer BN, but accepts CPN), **cpn-pref** (prefer CPN, but accepts BN), **none**, or leave blank. Leave blank when connected to another switch or to a network other than the AT&T Switched Network.

NOTE:
A 4-second delay occurs in terminating the call to the far-end station if the connecting switch does not respond to the request.

Night Serv

Specifies a night service extension (can be a VDN extension) per Service/Feature. An entry other than blank overrides Night Service entry on page 1 of the screen. This entry can be overridden by the Trunk/Member Night Service entry when provided. Valid entries are an assigned extension, the attendant group access code (**attd**), or leave blank.

Field Descriptions for the CBC Trunk Group Usage Allocation page

```
add trunk-group next                                                 Page y of x
                     CBC TRUNK GROUP USAGE ALLOCATION
Usage Allocation Plan 1     Usage Allocation Plan 2     Usage Allocation Plan 3
                Min# Max#                   Min# Max#                   Min# Max#
Service/Feature Chan Chan  Service/Feature Chan Chan  Service/Feature Chan Chan

_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
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_______________  __   __   _______________  __   __   _______________  __   __
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_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
_______________  __   __   _______________  __   __   _______________  __   __
```

Screen 142. CBC Trunk Group Usage Allocation

The CBC Trunk Group Usage Allocation screen sets a minimum and maximum number of members for up to ten different Services/Features for up to three different Usage Allocation Plans (1–3). Refer to ''Call-by-Call Service Selection'' on page 1513 for a detailed description of Usage Allocation Plans.

Service/Feature

Specifies the ISDN Services/Features that can be requested at call setup time when using this trunk group. See Service Type description for a list of predefined Services/Features that can be received on a call by call basis. In addition, the user defined service types can also be used. The identifier **other** is used for all Services/Features not explicitly specified.

Min# Chan

Indicates the minimum number of members of an ISDN trunk group with a Service Type field of **cbc** that a particular Service/Feature can use at any given time. The sum of the minimum number of members for all Service/Features must not exceed the total number of members of the trunk group. Valid values are **0-99** or blank.

Max# Chan

Indicates the maximum number of members of a ISDN trunk group with a Service Type field of **cbc** that a particular Service/Feature can use at any given time. This field must be completed if a Service/Feature has been entered in the Incoming Call Handling Treatment Table screen. Valid values are **0** to **99** or blank.

Field descriptions for the CBC Service Trunk Group Allocation Plan Assignment Schedule page

```
add trunk-group next                                                 Page y of x
            CBC SERVICE TRUNK GROUP ALLOCATION PLAN ASSIGNMENT SCHEDULE

  Usage Method:

         Fixed? y      Allocation Plan Number: 1
     Scheduled? n


Usage Allocation Plan Activation Schedule:

        Act  Plan    Act  Plan    Act  Plan    Act  Plan    Act  Plan    Act  Plan
        Time  #      Time  #      Time  #      Time  #      Time  #      Time  #
Sun     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Mon     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Tue     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Wed     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Thu     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Fri     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
Sat     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __     __:__ __
```

Screen 143. CBC Service Trunk Group Allocation Plan Assignment Schedule

The CBC Service Trunk Group Allocation Plan Assignment Schedule screen provides for administering a fixed schedule or administering a schedule that can change up to 6 times a day for each day of the week. This screen determines which CBC Service Trunk Group Allocation Plan will be in use at any given time.

Fixed

Indicates whether the allocation plan will be fixed. If **y** is entered, the plan number entered in the Allocation Plan Number field will be enabled.

Allocation Plan Number

Specifies the CBC Trunk Allocation Plan (1 through 3) that is in effect if a fixed usage method has been selected. This field must be assigned if the Fixed field is **y**. Valid entries are **1–3** or blank.

Scheduled

Indicates whether or not the allocation plans will be in effect according to the schedule found on this page. If **y** is entered in this field then there must be at least one entry in the schedule.

Act Time

Indicates the time the usage allocation plan administered in the next field (Plan #) will become effective. Enter the time in military time. There must be at least one entry per day. Valid entries are **00:00** through **23:59**.

Plan

Specifies the number of the usage allocation plan that will be in effect from the activation time until the activation time of the next scheduled plan change. Valid entries are **1** to **3** or blank.

Field descriptions for the Wideband Support Options page

```
add trunk-group next                                              Page y of x
                     Wideband Support Options

                             H0? n
                            H11? n
                            H12? n
                          NxDS0? y     Contiguous? n
```

Screen 144. Wideband Support Options

The Wideband Support Options screen appears immediately before the trunk group member pages. The actual page number will vary.

NOTE:
All B-channels that comprise the wideband call must reside on the same ISDN-PRI facility. Also, all trunk members in an ISDN trunk group with the Wideband Support field set to **y** must be from a TN464C (or later) circuit pack.

H0

Enter **y** to specify the ISDN information transfer rate for 384-kbps of data, which is comprised of six B-channels. When a trunk group is administered to support H0, the trunk/hunt algorithm to satisfy a call requiring 384-kbps of bandwidth uses a fixed allocation scheme.

H11

Enter **y** to specify the ISDN information transfer rate for 1536-kbps of data, which is comprised of 24 B-channels. When a trunk group is administered to support H11, the trunk/hunt algorithm to satisfy a call requiring 1536-kbps bandwidth uses a fixed allocation scheme.

H12

Enter **y** to specify the ISDN information transfer rate for 1920-kbps of data, which is comprised of 30 B-channels. When a trunk group is administered to support H12, the trunk/hunt algorithm to satisfy a call requiring 1920-kbps bandwidth uses a fixed allocation scheme.

NxDS0

Enter **y** to specify the “N by DS-zero” multi-rate service.

Contiguous

Specifies whether or not to hunt contiguous NXDS0 channels. This field only appears if the NxDS0 field is **y**.

The trunk/hunt algorithm to satisfy an NXDS0 call is as follows:

— Enter **y** to specify the "floating" scheme. NXDS0 calls are placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without constraint on the starting channel (no fixed starting point trunk).

NOTE:
H0 and NXDS0 "floating" scheme cannot both be **y**.

— Enter **n** to specify the "flexible" scheme. NXDS0 calls are placed on any set of B-channels on the same facility as long as the requested bandwidth is satisfied. There are no constraints such as contiguity of B-channels or fixed starting points.

Field Descriptions for the Group Member Assignments page

```
add trunk-group next                                         Page y of x
                               TRUNK GROUP
                               Administered Members(min/max): xxx/yyy
GROUP MEMBER ASSIGNMENTS          Total Administered Members: xxx

       Port      Code      Sfx  Name         Night   Sig Grp
    1: _______   _____      _   __________   _____     ___
    2: _______   _____      _   __________   _____     ___
    3: _______   _____      _   __________   _____     ___
    4: _______   _____      _   __________   _____     ___
    5: _______   _____      _   __________   _____     ___
    6: _______   _____      _   __________   _____     ___
    7: _______   _____      _   __________   _____     ___
    8: _______   _____      _   __________   _____     ___
    9: _______   _____      _   __________   _____     ___
   10: _______   _____      _   __________   _____     ___
   11: _______   _____      _   __________   _____     ___
   12: _______   _____      _   __________   _____     ___
   13: _______   _____      _   __________   _____     ___
   14: _______   _____      _   __________   _____     ___
   15: _______   _____      _   __________   _____     ___
```

Screen 145. ISDN Group Member Assignments

The total number of pages, and the first page of Group Member Assignments, will vary depending on whether the CBC and Wideband Support pages display.

NOTE:
When supporting DCS, Member Number Assignments must be the same between nodes (Member #1 must be Member #1 at the far-end trunk group).

Port

When using ISDN-BRI interfaces, B-channel 1 is the port number while B channel 2 is the port number plus 16. For example, if B channel 1's port number is 01A1002, then B channel 2's port number is 01A1018.

When using ISDN-PRI interfaces, the port number will be the one allied with the B-channel. For example, if the DS1 is located in 01A10, then B channel 1 will be 01A1001, B channel 2 will be 01A1002 and so forth.

Sig Grp

When using ISDN-BRI interfaces, this field becomes unadministrable. Otherwise, enter the appropriate signaling group number. If a DS1 interface appears in one signaling group, then the number of that signaling group appears as a default in the Sig Grp column for any trunk on that interface after the screen is submitted. This value cannot be altered without adding a DS1 interface to another signaling group. If a DS1 circuit pack appears in more than one signaling group, then the Sig Grp field cannot be left blank. One of the signaling group members must be entered. Valid values are **1–0** or **1–166** depending on the amount of system memory.

Related topics

Refer to ''ISDN service'' on page 1610 for detailed information about Integrated Services Digital Network trunks.

Refer to ''DS1 Trunk Service'' on page 1542 for detailed information on DS1.

ISDN Numbering — Private

This screen supports Private Numbering Plans (PNP). It allows you to specify the digits to be put in the Calling Number information element (IE), the Connected Number IE, and the QSIG Party Number for extensions in the Private Numbering Plan.

Avaya MultiVantage supports private-network numbers up to 15 digits long. If the total number — including the level 1 and 2 prefixes, the PBX identifier, and the extension — is more than 15 digits long, neither QSIG PartyNumbers nor the information elements are created or sent.

```
change isdn private-numbering                                   Page 1 of 1

                    ISDN NUMBERING - PRIVATE FORMAT


         Network Level: _                   PBX Identifier: ____
          Level 2 Code: _____               Deleted Digits: _
          Level 1 Code: _____
```

Screen 146. ISDN Number — Private Format

Network Level

Enter the value of the highest regional level employed by the PNP network. Use the following table to find the relationship between the network level and the Numbering Plan Identification/Type of Number (NPI/TON) encoding used in the QSIG PartyNumber or the Calling Number and Connected Number IEs.

Valid entries	Usage
0	NPI - PNP TON - local
1	NPI - PNP TON - Regional Level 1
2	NPI - PNP TON - Regional Level 2
blank	If this field is blank and the Send Calling Number and/or Send Connected Number field is **y** or **r** with **private** specified for the Numbering Format field on the ISDN Trunk Group screen, the Calling Number and/or Connected Number IEs will not be sent. If the field is left blank but **private** has been specified in the Numbering Format field on the ISDN Trunk Group screen, the Identification Number (PartyNumber data type) is sent for QSIG PartyNumbers encoded in ASN.1-defined APDUs. In this case, the ASN.1 data type containing the PartyNumber (PresentedAddressScreened, PresentedAddressUnscreened, PresentedNumberScreened, or PresentedNumberUnscreened) is sent marked as “PresentationRestricted” with “NULL” for the associated digits.

PBX Identifier

Valid entries	Usage
0 to **9**	Enter up to 5 digits. Digits entered in this field are added to the beginning of local extensions. The PBX identifier is added first, before Level 1 or Level 2 codes. For example, in private networks with a 7-digit numbering plan, the first 3 digits of the private-network number are often considered the PBX identifier.

NOTE:
The PBX identifier does not depend on the network level. Therefore, in a level 0 private network you can use this field to create a level 0 number longer than 5 digits.

Level 2 Code

Enter the switch's second level regional code in the network. Administer this field carefully. The system will not check to ensure you have entered a code that supports your entry in the Network Level field. You cannot enter anything in this field unless the Network Level field is set to **2**.

Valid entries	Usage
0 to **9**	Enter up to 5 digits.
blank	Because blank regional codes are valid, an entry is not required if the Network Level field is 2.

In QSIG standards, this level 2 code is called the Level 1 Regional Code.

Deleted Digits

In some networks, the leading digits of the number you dial internally to reach any extension are not part of its longer private network number. For example, extension 35581 might have a private network number of 221-5581. To send this network number correctly, you must set the Deleted Digits field to **1** (that is, tell the switch to delete 1 digit before inserting the PBX Identifier of 221).

Valid entries	Usage
0 **1** **2**	Enter the number of digits to be deleted from the most significant digits of a local extension prior to sending it in the Calling or Connected Number IEs or in the QSIG PartyNumber. This allows UDP network numbers to be converted to PNP Complete Numbers.

Level 1 Code

Enter the switch's first level regional code in the network. Administer this field carefully. The system will not check to ensure you have entered a code that supports your entry in the Network Level field. You cannot enter anything in this field unless the Network Level field is set to **1** or **2**.

Valid entries	Usage
0 to **9**	Enter up to 5 digits.
blank	Because blank regional codes are valid, an entry is not required if the Network Level field is 1 or 2.

In QSIG standards, this level 1 code is called the Level 0 Regional Code.

ISDN Numbering — Public/ Unknown

This screen supports the ISDN Call Identification Display feature. The feature provides a name/number display for display-equipped stations within an ISDN network. The system uses the caller's name and number and displays it on the called party's display. Likewise, the called party's name and number can be displayed on the caller's display.

The screen allows you to specify the desired digits for the Calling Number IE and the Connected Number IE (in addition to the QSIG Party Number) for any extension in the Public and/or Unknown Number Plans.

Administer these screens if either the Send Calling Number, Send Connected Number field is specified, or the Supplementary Service Protocol field is **b** on the trunk group screen.

NOTE:
If the table is not properly administered and the Send Calling Number or Send Connected Number field is **y** or **r** and the Numbering Format field on the ISDN Trunk Group screen is **public** or **unknown**, the Calling Number and Connected Number IE are not sent. If the table is not administered, but the Send Calling Number or Send Connected Number field is **public** or **unknown**, the Identification Number (PartyNumber data type) is not sent for QSIG PartyNumbers. In this case, the ASN.1 data type containing the PartyNumber (PresentedAddressScreened, PresentedAddressUnscreened, PresentedNumberScreened, or PresentedNumberUnscreened) will be sent marked as "PresentationRestricted" with "NULL" for the associated digits.

```
change isdn public-unknown-numbering                        Page 1 of 8
                    ISDN NUMBERING - PUBLIC/UNKNOWN FORMAT

                                        Total                                  Total
Ext Ext      Trk       CPN               CPN Ext Ext      Trk       CPN             CPN
Len Code     Grp(s)    Prefix            Len Len Code     Grp(s)    Prefix          Len

 5  4_______ ______    73283___________  10   _ ________ ______    _______________  __
 7  3646180_ ______    210_____________  10   _ ________ ______    _______________  __
 _  ________ ______    ________________  __   _ ________ ______    _______________  __
 _  ________ ______    ________________  __   _ ________ ______    _______________  __
 _  ________ ______    ________________  __   _ ________ ______    _______________  __
 _  ________ ______    ________________  __   _ ________ ______    _______________  __
 _  ________ ______    ________________  __   _ ________ ______    _______________  __
 _  ________ ______    ________________  __   _ ________ ______    _______________  __
 _  ________ ______    ________________  __   _ ________ ______    _______________  __
 _  ________ ______    ________________  __   _ ________ ______    _______________  __
 _  ________ ______    ________________  __   _ ________ ______    _______________  __
 _  ________ ______    ________________  __   _ ________ ______    _______________  __
 _  ________ ______    ________________  __   _ ________ ______    _______________  __
```

Screen 147. ISDN Numbering Public/Unknown

Ext Len

Specifies the number of digits the extension can have.

Valid entries	Usage
0 to **7**	Corresponds to the extension lengths allowed by the dial plan.

Ext Code

Allows for groups of extensions to be administered.

NOTE:
When **0** alone is entered, the Ext Len field must be 1 and the DDD number must be 10-digits.

Valid entries	Usage
0 to **9**	The Ext Code can be up to 7-digits long depending on the Ext Len field entry. The entry cannot be greater than the Ext Len field entry. For example, in the case of a 4-digit Ext Len field entry, an Ext Code of 12 is the equivalent of all extensions of the screen 12xx, excluding any explicitly listed longer codes. If a code of 123 is also listed, the 12 code is equivalent of all extensions of the screen 12xx except extensions of the screen 123x. The coding precludes having to list all the applicable 12xx extensions.
attd	For attendant

Trk Grp(s)

The switch generates the station's identification number if there is an entry in the Ext Code field, and this field is administered with the trunk group number carrying the call.

Valid entries	Usage
1 to 7 digits	Enter the valid administered ISDN trunk-group number or a range of group numbers. For example, if trunk groups 10 through 24 use the same CPN Prefix, enter **10-24**.
blank	The identification numbers are not dependent on which trunk group the call is carried.

CPN Prefix

Use this field to specify the number that is added to the beginning of the extension to form a Calling or Connected Number.

Valid entries	Usage
1 to 15 digits	Only digits are allowed in the CPN Prefix column. Leading spaces, or spaces in between the digits, are not allowed. ■ If the length of the CPN Prefix matches the Total CPN Length, the extension number is not used to formulate the CPN number. ■ If the number of digits in the CPN Prefix plus the extension length exceeds the administered Total CPN Length, excess leading digits of the extension are deleted when formulating the CPN number. ■ If the number of CPN Prefix digits plus the extension length is less than the Total CPN Length, the entry is not allowed. ■ If the Total CPN Length is 0, no calling party number information is provided to the called party and no connected party number information is provided to the calling party.
blank	If this field is blank, the extension is sent unchanged. This is useful in countries where the public network is able to insert the appropriate CPN Prefix to form an external DID number.

Total CPN Len

Valid entries	Usage
0 to **15**	Enter the total number of digits to send.

ISDN-BRI Trunk Circuit Pack

This screen administers an ISDN-BRI circuit pack. See *Avaya MultiVantage™ Hardware Solutions Guide* for information on the maximum number of ISDN-BRI circuit packs that you can administer.

```
change bri-trunk-board                                            Page 1 of 1
                         ISDN-BRI TRUNK CIRCUIT PACK

              Location: 01A09                          Name:  ______________
  Interface Companding: a-law_  DCP/Analog Bearer Capability: 3.1kHz
 T3 Timer Length (sec): 15_


Port  Interface   Side  Cntry/Peer TEI  Time  Invert Synch   Layer 1  Detect
                         Protocol       Fill  Bits?  Source? Stable?  Slips?
  1:  user_______         12__     0__  flags  n       n       n       n
  2:  network____         etsi     0__  ones_  y       y       y       y
  3:  user_______         2___     auto flags  y       n       y       n
  4:  peer-slave_  b      QSIG     0__  flags  n       y       y       n
  5:  peer-master  a      QSIG     auto ones_  n       n       n       n
  6:  ___________         ____     0__  ones_  n       n       y       n
  7.  ___________         ____     0__  ones_  n       n       y       n
  8:  ___________         ____     0__  ones_  n       n       y       n
```

Screen 148. BRI Trunk (using a TN2185 circuit pack)

Location

This is a display-only field. It shows the TN2185 circuit pack location (PPCSS)

Name

Valid entries	Usage
1–15 alpha-numeric characters	This name is used to identify the circuit pack.

Interface Companding

Valid entries	Usage
a-law **mu-law**	Indicates the companding algorithm expected by the system at the far end.

DCP/Analog Bearer Capability

Valid entries	Usage
3.1kHz **speech**	Indicates how to encode the Bearer Capability IE for an outgoing call originated by a DCP or analog endpoint.

T3 Timer Length (sec)

Valid entries	Usage
1 to **127**	Tells the TE side how long to wait, in seconds, for an inactive Layer 1 to become active.

Port

This is a display-only field. It shows the port number to which parameters administered on the row apply.

Interface

Valid entries	Usage
network **user** **peer-master** **peer-slave**	Tells call processing software whether a particular port is connected to a user/network or a peer interface. These entries are valid for the TN2185. You can enter **peer-slave** only if the QSIG Basic Call Setup feature is enabled

Side

Valid entries	Usage
a **b**	Determines how glare conditions are handled when Interface field is **peer-slave**.

Cntry/Peer Protocol

Tells call processing software which ISDN protocol standard is applied.

Valid entries	Usage
1 to **25** **etsi**	When this field is **10**, **12**, **13**, or **etsi**, the Protocol Version field is equivalent to **b** on the DS1 circuit pack screen. When the Cntry/Peer Protocol field is set **10**, **12**, **13**, or **etsi**, set the Provol Version field to **b**. For all other administered values, the Protocol Version sets to ***a***.
QSIG	When the Interface field is **peer-slave** or **peer-master**, this field must be **QSIG**. The choice **QSIG** is valid only when the Interface field is **peer-slave**.
blank	You cannot leave this field blank if the Interface field is set to a valid, non-blank value

TEI

Valid entries	Usage
auto	
0	

Time Fill

The bit pattern sent on the D-channel between valid LAPD packets.

Valid entries	Usage
flags	
ones	

Invert Bits

Valid entries	Usage
y/n	

Synch Source

Valid entries	Usage
y	
n	

Layer 1 Stable

Valid entries	Usage
y **n**	Tells call processing and maintenance software whether to expect the network to drop Layer 1 when the BRI port is idle. Only the TN2185 may be set to **n**.

Detect Slips

Valid entries	Usage
y/n	Tells maintenance software whether slips reported by the BRI port should be logged.

Field descriptions for page 1

```
change bri-trunk-board                                          Page 1 of 1
                         ISDN-BRI TRUNK CIRCUIT PACK

              Location: 01A09                  Name:  ______________
   Interface Companding: a-law_  DCP/Analog Bearer Capability: 3.1kHz


Port  Interface   Side  Cntry/Peer TEI  Time Invert
                        Protocol        Fill  Bits?
  1:  network__           12__     0__  flags   n
  2:  network__           etsi     0__  ones_   y
  3:  network__           2___     auto flags   y
  4:  peer-master  b      QSIG     0__  flags   n
  5:  peer-master  a      QSIG     auto ones_   n
  6:  __________          ____     0__  ones_   n
  7.  __________          ____     0__  ones_   n
  8:  __________          ____     0__  ones_   n
  9:  __________          ____     0__  ones_   n
 10:  __________          ____     0__  ones_   n
 11:  __________          ____     0__  ones_   n
 12:  __________          ____     0__  ones_   n
```

Screen 149. BRI Trunk (with a TN556B or TN2198 circuit pack)

The following field descriptions are unique to the ISDN-BRI Circuit Pack screen with a TN556B or TN2198 circuit pack. The following fields do not display with a TN556B or TN2198 circuit pack:

- T3 Timer Length (sec)
- Synch Source
- Layer 1 Stable
- Detect Slips

Interface

Valid entries	Usage
network **peer-master**	Tells call processing software whether a particular port is connected to a user/network or a peer interface. These entries are valid for the TN556B. You can enter **peer-master** only if the QSIG Basic Call Setup feature is enabled

Side

Valid entries	Usage
a **b**	Determines how glare conditions are handled when Interface field is peer-slave. This field is not administrable when the Interface field is **network**.

Cntry/Peer Protocol

Tells call processing software which ISDN protocol standard is applied.

Valid entries	Usage
1 to **25**	When this field is **10**, **12**, or **13**, the Protocol Version field is equivalent to **b** on the DS1 circuit pack screen.
etsi	When this field is **etsi**, the Protocol Version field is equivalent to **b** on the DS1 circuit pack screen.
QSIG	When the Interface field is **peer-master**, this field must be **QSIG**.
blank	You cannot leave this field blank if the Interface field is set to a valid, non-blank value.

Field descriptions for page 2

NOTE:
If administering a TN2185 circuit pack, 8 ports appear; otherwise, 12 ports appear.

```
change bri-trunk-board                                         Page 2 of 2
                                ISDN-BRI TRUNK CIRCUIT PACK

 Port  Interwork  XID  Endpt    SPID      Endpt    SPID      Endpt  Max
       Message   Test? Init?               ID                 ID   NCA TSC
   1:  PROGress   y      n   ____________ __  ____________  __     0___
   2:  ALERTing   y      y   908957200000 __  ____________  __     0___
   3:  PROGress   y      y   0001________ __  ____________  __     0___
   4:  PROGress   n      n   ____________ __  ____________  __     0___
   5:  PROGress   n      y   625761449___ 01  ____________  __     0___
   6:  PROGress   n      n   ____________ __  ____________  __     0___
   7.  PROGress   n      n   ____________ __  ____________  __     0___
   8:  PROGress   n      n   ____________ __  ____________  __     0___

   Port Directory   Directory         Port Directory   Directory
          Number      Number                 Number      Number
     1:                                 5:
     2:                                 6:
     3:                                 7:
     4:                                 8:
```

Screen 150. BRI Trunk - Page 2 (using a TN2185 circuit pack)

```
change bri-trunk-board                                         Page 2 of 2
                                ISDN-BRI TRUNK CIRCUIT PACK

 Port  Interwork  XID  Endpt    SPID      Endpt                      Max
       Message   Test? Init?               ID                      NCA TSC
   1:  PROGress   n      y   ____________ __                           0___
   2:  ALERTing   n      y   ____________ __                           0___
   3:  PROGress   n      y   ____________ __                           0___
   4:  PROGress   n      y   ____________ __                           0___
   5:  PROGress   n      y   ____________ __                           0___
   6:  PROGress   n      y   ____________ __                           0___
   7.  PROGress   n      y   ____________ __                           0___
   8:  PROGress   n      y   ____________ __                           0___
   9:  PROGress   n      y   ____________ __                           0___
  10:  PROGress   n      y   ____________ __                           0___
  11:  PROGress   n      y   ____________ __                           0___
  12:  PROGress   n      y   ____________ __                           0___
```

Screen 151. BRI Trunk - Page 2 (using a TN2198/TN556B circuit pack)

NOTE:
You cannot change the Endpt Init, SPID, or Endpt ID port parameters unless that port is busied out or unadministered. It is possible to change all other fields on this page even if the corresponding port is active.

NOTE:
If the Interface field on page 1 contains a valid value when the screen is submitted, the contents of the fields on page 2 for that port are validated. If the Interface field is blank when the screen is submitted, the fields on this page for that port reset to their default values.

Port

This is a display-only field. It shows the port number to which parameters administered on the row apply.

Interworking Message

This field determines what message the switch sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group).

Valid entries	Usage
PROGress	Normally select this value. PROGress asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk.
ALERTing	ALERTing causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

XID Test

Valid entries	Usage
y/n	Indicates whether the far end supports the Layer 2 XID test.

Endpt Init

Indicates whether the far end supports endpoint initialization. Avaya MultiVantage blocks you from changing this field unless the port is busied out or unadministered.

Valid entries	Usage
y	If set to **y**, the SPID field must *not* be blank. Avaya MultiVantage blocks you from changing this field and the SPID field unless that port is busied out or unadministered.
n	If set to **n**, the SPID and Endpt ID fields must be blank.

SPID

A 12-digit field containing the SPID expected by the far end. Avaya MultiVantage blocks you from changing this field unless the port is busied out or unadministered. The only protocol supported for SPID initialization is Bellcore (Country Code 1). Trunks will not be put in service if SPID installation is not successful.

Valid entries	Usage
Any string of 1 to 12 digits	Leading zeroes considered significant and not ignored

Endpt ID

A 2-digit field containing the Endpoint Identifier expected by the far end. Avaya MultiVantage blocks you from changing this field unless the port is busied out or unadministered.

Valid entries	Usage
00 to **62**	Leading zeroes considered significant and not ignored.

Max NCA TSC

Valid entries	Usage
0 through **63**.	This 2-digit field gives the maximum number of Non-Call-Associated Temporary Signaling Connections allowed on this BRI D-channel. This field's function is the same as the field with the same name on the Signaling Group screen.

Port

This is a display-only field. It shows the port number to which parameters administered on the row apply.

Directory Number

These 10-digit fields contain the directory numbers assigned to the interface, which it views as being allocated to 2 separate endpoints.

Valid entries	Usage
Any string of 1 to 10 digits	These fields must be administered in pairs. That is, if you enter a value in one field, you must enter a value in the other.

Trunk Member Administration

Administer BRI trunk members using the following scheme to address the individual B-channels:

- B-channel 1 uses the port address of the BRI Trunk Port.
- B-channel 2 uses the port address of B-channel 1 incremented by 16.

When adding a BRI trunk to an ISDN trunk-group, Avaya MultiVantage blocks you from administering a Signaling Group for that trunk member.

Avaya MultiVantage blocks you from administering a BRI trunk member if the port has not yet been administered on the BRI Trunk screen.

For example, administer the B-channels on a TN2185 circuit pack inserted in slot 01A10 as follows:

Port	B-channel 1	B-channel 2
1	01A1001	01A1017
2	01A1002	01A1018
3	01A1003	01A1019
4	01A1004	01A1020
5	01A1005	01A1021
6	01A1006	01A1022
7	01A1007	01A1023
8	01A1008	01A1024

Interactions

The **add bri-trunk board PPCSS** command is rejected if PPCSS identifies a TN556B circuit pack, and a port on that circuit pack has already been assigned to a station or data-module. If a TN556B circuit pack has been administered as a BRI trunk circuit pack, any port on that circuit pack is prevented from being assigned to a station or data-module.

Language Translations

When you want to use a language other than English, French, Italian, and Spanish to display messages on your display phones, use the Language Translation screens to define the messages. Set up your messages using Roman letters, punctuation, digits and blank spaces. Diacritical marks are not supported.

NOTE:
If “user-defined” is entered for the display language on the station screen or attendant screen, and no messages are defined on these screens, a string of asterisks appears on all display messages.

In this section, the field descriptions are listed before the screens

English

This is a display-only field. It contains the English version of the message on the display.

Meaning of English term

This is a display-only field. It explains the abbreviated English message.

Translation

Enter the message you want to appear on the phone display in place of the English message. Remember that a long message may be shortened on phones that display fewer than 32 characters.

```
change display-messages ad-programming                      Page  1 of  1
                              LANGUAGE TRANSLATIONS

     English                            Translation

  1. Press button to program.         1. ****************************************
  2. Change program?                  2. ***********************
  3. Yes = 1 No = 2                   3. ****************
  4. Enter number:                    4. ****************************************
  5. Press # to save.                 5. ****************
  6. Number saved.                    6. ****************************************
  7. Change label?                    7. ***********************
  8. Enter label:                     8. ****************************************
  9. Press * to advance; # to save.   9. ******************************
 10. Press * to reenter; # to save.  10. ******************************
 11. Label saved. Hang up to update. 11. ****************************************
```

Screen 152. Language Translations — AD programming

```
change display-messages auto-wakeup-dn-dst                      Page  1 of  3
                              LANGUAGE TRANSLATIONS

   1.     English:    AUTO WAKEUP - Ext:
      Translation: ********************:

   2.     English: WAKEUP ENTRY DENIED
      Translation: ************************

   3.     English: WAKEUP REQUEST CANCELED
      Translation: ****************************************

   4.     English: WAKEUP REQUEST CONFIRMED
      Translation: ****************************************

   5.     English: Wakeup Call
      Translation: **************************************

   6.     English:  Time:
      Translation: *****:
```

Screen 153. Language Translations — Auto-Wakeup-Dn-Dst

```
change display-messages auto-wakeup-dn-dst                      Page  2 of  3
                              LANGUAGE TRANSLATIONS

   7.     English:    DO NOT DIST - Ext:
      Translation: ********************:

   8.     English:     DO NOT DIST - Group:
      Translation: ***********************:

   9.     English: DO NOT DIST ENTRY DENIED
      Translation: ************************

  10.     English: THANK YOU - DO NOT DIST ENTRY CONFIRMED
      Translation: ****************************************

  11.     English: THANK YOU - DO NOT DIST REQUEST CANCELED
      Translation: ****************************************
```

Screen 154. Language Translations — Auto-Wakeup-Dn-Dst

```
change display-messages auto-wakeup-dn-dst                       Page  3 of  3
                             LANGUAGE TRANSLATIONS

  12.     English: INTERVAL FULL
      Translation: *************

  13.     English: NO PERMISSION
      Translation: *************

  14.     English: SYSTEM FULL
      Translation: *************

  15.     English: TOO SOON
      Translation: *************

  16.     English: INVALID EXTENSION - TRY AGAIN
      Translation: ****************************************

  17.     English: INVALID GROUP - TRY AGAIN
      Translation: ****************************************
```

Screen 155. Language Translations — Auto-Wakeup-Dn-Dst

```
change display-messages call identifiers                         Page  1 of  4
                              LANGUAGE TRANSLATIONS

     English               Meaning of English term                  Translated
     Term                                                           Term

  1. sa            ACD Supervisor Assistance                     1: **
  2. ac            Attendant Assistance Call                     2: **
  3. tc            Attendant Control Of A Trunk Group            3: **
  4. an            Attendant No Answer                           4: **
  5. pc            Attendant Personal Call                       5: **
  6. rc            Attendant Recall Call                         6: **
  7. rt            Attendant Return Call                         7: **
  8. sc            Attendant Serial Call                         8: **
  9. co            Controlled Outward Restriction                9: **
 10. cs            Controlled Station To Station Restriction    10: **
 11. ct            Controlled Termination Restriction           11: **
 12. db            DID Find Busy Station With CO Tones          12: **
 13. da            DID Recall Go To Attendant                   13: **
 14. qf            Emergency Queue Full Redirection             14: **
 15. hc            Held Call Timed Reminder                     15: **
```

Screen 156. Language Translations — Call-Identifiers

```
change display-messages call identifiers                        Page  2 of  4
                              LANGUAGE TRANSLATIONS

     English                 Meaning of English term                Translated
     Term                                                           Term

 16. ic             Intercept                                    16: **
 17. ip             Interposition Call                           17: **
 18. ld             LDN Calls On DID Trunks                      18: **
 19. so             Service Observing                            19: **
 20. na             Unanswered Or Incomplete DID Call            20: **
 21. ACB            Automatic Callback                           21: ********
 22. callback       Callback Call                                22: ********
 23. park           Call Park                                    23: ********
 24. control        Control                                      24: ********
 25. ICOM           Intercom Call                                25: ********
 26. OTQ            Outgoing Trunk Queuing                       26: ********
 27. priority       Priority Call                                27: ********
 28. recall         Recall Call                                  28: ********
 29. return         Return Call                                  29: ********
 30. ARS            Automatic Route Selection                    30: ********
```

Screen 157. Language Translations — Call-Identifiers

```
change display-messages call identifiers                        Page  3 of  4
                              LANGUAGE TRANSLATIONS

     English                 Meaning of English term                Translated
     Term                                                           Term

 31. forward        Call Forwarding                              31: ********
 32. cover          Cover                                        32: ********
 33. DND            Do Not Disturb                               33: ********
 34. p              Call Pickup                                  34: *
 35. c              Cover All Calls                              35: *
 36. n              Night Station Service, Including No Answer   36: *
 37. B              All Calls Busy                               37: *
 38. f              Call Forwarding                              38: *
 39. b              Cover Busy                                   39: *
 40. d              Cover Don't Answer                           40: *
 41. s              Send All Calls                               41: *
 42. to             <calling party> to <called party>            42: **
 43. VDN            Vector Directory Number                      43: ***
 44. hunt           Station Hunting, Origination                 44: ********
 45. h              Station Hunting, Termination                 45: *
```

Screen 158. Language Translations — Call-Identifiers

```
change display-messages call identifiers                          Page  4 of  4
                              LANGUAGE TRANSLATIONS

     English         Meaning of English term                    Translated
     Term                                                       Term

 46. OPERATOR        Operator                                 46: ***************
 47. EXT             Extension                                47: ***************
 48. OUTSIDE CALL    Outside Call                             48: ***************
 49. UNKNOWN NAME    Unknown Name                             49: ***************
 50. CONFERENCE      Conference                               50: ************
 51. ringing         Ringing                                  51: ********
 52. busy            Busy                                     52: ********
 53. busy(I)         Busy With Intrusion Allowed              53: ********
 54. wait            Wait                                     54: *****
 55. (I)             Intrusion                                55: ***
 56. Sta             Station                                  56: *********
 57. Trk             Trunk                                    57: *********
 58: offered         QSIG call offered to remote endpoint     58: ********
 59: cl              Controlled Toll Restriction              59: **
 60: vm              Call to Attendant Out of Voicemail       60: **
```

Screen 159. Language Translations — Call-Identifiers

```
change display-messages date-time                                 Page  1 of  1
                              LANGUAGE TRANSLATIONS

     English           Translation            English          Translation

  1. SUNDAY         1: ***********        11. APRIL         11: ***********
  2. MONDAY         2: ***********        12. MAY           12: ***********
  3. TUESDAY        3: ***********        13. JUNE          13: ***********
  4. WEDNESDAY      4: ***********        14. JULY          14: ***********
  5. THURSDAY       5: ***********        15. AUGUST        15: ***********
  6. FRIDAY         6: ***********        16. SEPTEMBER     16: ***********
  7. SATURDAY       7: ***********        17. OCTOBER       17: ***********
  8. JANUARY        8: ***********        18. NOVEMBER      18: ***********
  9. FEBRUARY       9: ***********        19. DECEMBER      19: ***********
 10. MARCH         10: ***********

 20.     English: SORRY, TIME UNAVAILABLE NOW
     Translation: ****************************************
```

Screen 160. Language Translations — Date-Time

```
change display-messages leave-word-calling                          Page  1 of  2
                              LANGUAGE TRANSLATIONS

   1.     English:             MESSAGES FOR
      Translation: ************************

   2.     English: WHOSE MESSAGES? (DIAL EXTENSION NUMBER)
      Translation: ****************************************

   3.     English: END OF MESSAGES (NEXT TO REPEAT)
      Translation: ****************************************

   4.     English: MESSAGES UNAVAILABLE - TRY LATER
      Translation: ****************************************

   5.     English: MESSAGE RETRIEVAL DENIED
      Translation: ****************************************

   6.     English: MESSAGE RETRIEVAL LOCKED
      Translation: ****************************************
```

Screen 161. Language Translations — Leave-Word-Calling

```
change display-messages leave-word-calling                          Page  2 of  2
                              LANGUAGE TRANSLATIONS

   7.     English: NO MESSAGES
      Translation: ****************************************

   8.     English: IN PROGRESS
      Translation: ****************************************

   9.     English: DELETED
      Translation: ****************************************

  10.     English: GET DIAL TONE, PUSH Cover Msg Retrieval
      Translation: ****************************************

  11.     English: Message Center (AUDIX) CALL
      Translation: ****************************************

  12.     English: CANNOT BE DELETED - CALL MESSAGE CENTER
      Translation: ****************************************
```

Screen 162. Language Translations — Leave-Word-Calling

```
change display-messages mailcious-call-trace                    Page  1 of  2
                              LANGUAGE TRANSLATIONS

   1.     English: MALICIOUS CALL TRACE REQUEST
      Translation: ****************************************

   2.     English: END OF TRACE INFORMATION
      Translation: ****************************************

  3.     English: original call redirected from:
     Translation: *****************************:

  4.     English:           voice recorder port:
     Translation: *****************************:

  5.     English:  MCT activated by:        for:
     Translation: *****************:       ****:
```

Screen 163. Language Translations — Malicious-Call-Trace

```
change display-messages mailcious-call-trace                   Page  2 of  2
                             LANGUAGE TRANSLATIONS


  6.     English:    party :                    (EXTENSION)
     Translation: ******** :       **********************

  7.     English:    party :                 (ISDN SID/CNI)
     Translation: ******** :              ****************

  8.     English:    party :                      (PORT ID)
     Translation: ******** :           *******************

  9.     English:    party :                 (ISDN PORT ID)
     Translation: ******** :           *******************
```

Screen 164. Language Translations — Malicious-Call-Trace

```
change display-messages miscellaneous-features                   Page  1 of 7
                              LANGUAGE TRANSLATIONS

     English                                 Translation

   1. ALL MADE BUSY                       1: ***************
   2. BRIDGED                             2: ***************
   3. DENIED                              3: ***************
   4. INVALID                             4: ***************
   5. NO MEMBER                           5: ***************
   6. OUT OF SERVICE                      6: ***************
   7. RESTRICTED                          7: ***************
   8. TERMINATED                          8: ***************
   9. TRUNK SEIZED                        9: ***************
  10. VERIFIED                           10: ***************
  11. CDR OVERLOAD                       11: ***************
  12. ANSWERED BY                        12: ***************
  13. CALL FROM                          13: ***************
  14. Skills                             14: ***************
```

Screen 165. Language Translations — Miscellaneous-Features

```
change display-messages miscellaneous-features                  Page  2 of 7
                              LANGUAGE TRANSLATIONS
     English                Meaning of English term                Translated
     Term                                                          Term

  15. TOLL          Toll                                       15: ****
  16. FULL          Full                                       16: ****
  17. NONE          None                                       17: ****
  18. ORIG          Origination                                18: ****
  19. OTWD          Outward                                    19: ****
  20. CALL          <call> This Number                         20: ****
  21. INTL          International                              21: ****
  22. Info          Information                                22: *****
  23. p             Primary                                    23: *
  24. s             Secondary                                  24: *
  25. m             Mark                                       25: *
  26. p             Pause                                      26: *
  27. s             Suppress                                   27: *
  28. w             Wait For A Specified Time                  28: *
  29. W             Wait For Off-Premise Dial Tone             29: *
```

Screen 166. Language Translations — Miscellaneous Features

```
change display-messages miscellaneous-features                  Page 3 of 7

                              LANGUAGE TRANSLATIONS

  30.     English: You have adjunct messages
      Translation: ****************************************

  31.     English: Login Violation
      Translation: ****************************************

  32.     English: Barrier Code Violation
      Translation: ****************************************

  33.     English: Authorization Code Violation
      Translation: ****************************************

  34.     English: DIRECTORY - PLEASE ENTER NAME
      Translation: ****************************************

  35.     English: DIRECTORY UNAVAILABLE - TRY LATER
      Translation: ****************************************
```

Screen 167. Language Translations — Miscellaneous-Features

```
change display-messages miscellaneous-features                  Page 4 of 7

                              LANGUAGE TRANSLATIONS

  36.     English: NO MATCH - TRY AGAIN
      Translation: ****************************************

  37.     English: NO NUMBER STORED
      Translation: ****************************************

  38.     English: TRY AGAIN
      Translation: ****************************************

  39.     English: Ext          in EMRG Q
      Translation: ***          **********

  40.     English:       HUNT GROUP   NOT ADMINISTERED
      Translation: ****************   *********************

  41.     English: Q-time       calls
      Translation: ******       *****
```

Screen 168. Language Translations — Miscellaneous-Features

```
change display-messages miscellaneous-features                    Page 5 of 7
                               LANGUAGE TRANSLATIONS

   42.     English: Add Skill: Enter number, then # sign
       Translation: ****************************************

   43.     English: Remove Skill: Enter number, then # sign
       Translation: ****************************************

   44.     English: Enter Skill Level, then # sign
       Translation: ****************************************

   45.     English: Enter Agent LoginID
       Translation: ****************************************

   46.     English: Call Type
       Translation: ***************

   47.     English: Call Charge
       Translation: ********************
```

Screen 169. Language Translations — Miscellaneous-Features

```
change display-messages miscellaneous-features                   Page 6 of 7
                                 LANGUAGE TRANSLATIONS

   48.     English: Station Security Code Violation
       Translation: ***************************************
   49.     English: ENTER REASON CODE
       Translation: ***************************************
   50.     English: Whisper From
       Translation: ********************
   51.     English: Whisper To
       Translation: ********************
   52.     English: Press button to remove.
       Translation: ***************************************
   53.     English: Press # to remove.
       Translation: ********************
```

Screen 170. Language Translations — Miscellaneous-Features

```
change display-messages miscellaneous-features                     Page 6 of 7
                          LANGUAGE TRANSLATIONS

  54.      English: Button removed.
      Translation: ***************************************
  55.      English: Ringer On
      Translation: ***************************************
  56.      English: Ringer Off
      Translation: ***************************************
  57.      English: Ringer Abbreviated
      Translation: ***************************************
  58.      English: Ringer Delayed
      Translation: ***************************************
  59.      English: Select a held party's line to talk.
      Translation: ***************************************
```

Screen 171. Language Translations — Miscellaneous-Features

```
change display-messages miscellaneous-features                     Page 7 of 7
                          LANGUAGE TRANSLATIONS

  48.      English: Station Security Code Violation
      Translation: ***************************************
  49.      English: ENTER REASON CODE
      Translation: ***************************************
  50.      English: Whisper From
      Translation: ********************
  51.      English: Whisper To
      Translation: ********************
  52.      English: Press button to remove.
      Translation: ***************************************
  53.      English: Press # to remove.
      Translation: ********************
```

Screen 172. Language Translations — Miscellaneous-Features

```
change display-messages property-management                         Page  2 of  6
                              LANGUAGE TRANSLATIONS

7.     English: CHECK OUT FAILED
      Translation: ****************************************

   8.     English: MESSAGE NOTIFICATION FAILED
      Translation: ****************************************

   9.     English:    MESSAGE NOTIFICATION ON - Ext:
      Translation: ********************************:

  10.     English:   MESSAGE NOTIFICATION OFF - Ext:
      Translation: ********************************:
  11.     English: CHECK OUT COMPLETE: MESSAGE LAMP OFF
      Translation: ****************************************

  12.     English: CHECK OUT COMPLETE: MESSAGE LAMP ON
      Translation: ****************************************
```

Screen 173. Language Translations — Property-Management

```
change display-messages property-management                      Page  3 of  6
                              LANGUAGE TRANSLATIONS


 13.     English: MESSAGE LAMP ON
     Translation: ****************************************

 14.     English: MESSAGE LAMP OFF
     Translation: ****************************************
 15.     English: Occupied Rooms
     Translation: ****************************************
 16.     English: Enter Room Status (1-6)
     Translation: ****************************************
 17.     English: State, Enter number from 1 - 6
     Translation: ****************************************
 18.     English: DID
     Translation: *********************
```

Screen 174. Language Translations — Property-Management

```
change display-messages property-management                    Page 4 of 6
                              LANGUAGE TRANSLATIONS
   19.     English: DID VIEW: EXT?
       Translation: ****************************************
   20.     English: DID=                  CHANGE?
       Translation: ****                  **********************
   21.     English: DID VIEW DONE
       Translation: ****************************************
   22.     English: NO DID AVAILABLE
       Translation: ****************************************
   23.     English: CHECK IN COMPLETE, DID=
       Translation: *********************************
```

Screen 175. Language Translations — Property-Management

```
change display-messages property-management                   Page 5 of 6

                               LANGUAGE TRANSLATIONS
   24.     English: REMOVE(1), REPLACE(2)?
       Translation: ****************************************
   25.     English: DID REMOVED
       Translation: ****************************************
   26.     English: VIP CHECK IN - Ext:
       Translation: ****************************************
   27.     English: SPECIFY VID DID:
       Translation: ****************************************
   28.     English: CHECK IN COMPLETE, INVALID DID
       Translation: ****************************************
   29.     English: CHECK IN COMPLETE, DID UNAVAILABLE
       Translation: ****************************************
```

Screen 176. Language Translations — Property-Management

```
change display-messages property-management                   Page 6 of 6

                               LANGUAGE TRANSLATIONS
   30.     English: DID REMOVE - Ext:
       Translation: ****************************************
   31.     English: DID CHANGED
       Translation: ****************************************
   32.     English: AUTOMATIC ASSIGN(1), SELECT(2)?
       Translation: ****************************************
   33.     English: DID UNAVAILABLE
       Translation: ****************************************
```

Screen 177. Language Translations — Property-Management

```
change display-messages view-buttons                            Page 1 Of 1
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Analog Bridge Appearance       1. ************************
  2. Abbreviated Dial Program       2. ************************
  3. Abbrev Dial Character          3. ************************
  4. Abbreviated Dial               4. ************************
  5. Abrv/Delayed Ring Change       5. ************************
  6. Auto Circuit Assurance         6. ************************
  7. Admin Connection Alarm         7. ************************
  8. CDR Account Code               8. ************************
  9. TSA Administration Mode        9. ************************
 10. ACD After Call Work           10. ************************
 11. ACD Change Alert              11. ************************
 12. Alternate FRL                 12. ************************
 13. Supervisor Assist             13. ************************
 14. SVN Auth Code Halt            14. ************************
 15. Attendant Queue Calls         15. ************************
 16. Attendant Queue Time          16. ************************
```

Screen 178. Language Translations — View-buttons (page 1)

```
change display-messages view-buttons                            Page 2 Of 2
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Auto Message Waiting           1. ************************
  2. Automatic Call Back            2. ************************
  3. Automatic Dialing              3. ************************
  4. Automatic Intercom             4. ************************
  5. Auto-In Work Mode              5. ************************
  6. Automatic Wakeup               6. ************************
  7. Auxiliary Work Mode            7. ************************
  8. Bridged Appearance             8. ************************
  9. Busy Indicator                 9. ************************
 10. Call Appearance               10. ************************
 11. Call Displayed Number         11. ************************
 12. Call Forwarding               12. ************************
 13. Call Park                     13. ************************
 14. Call Pickup                   14. ************************
 15. Caller Information            15. ************************
 16. CAS (Branch) Backup           16. ************************
```

Screen 179. Language Translations — View-buttons (page 2)

```
change display-messages view-buttons                            Page 3 Of 3
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. CDR 1st Printer Alarm         1. ************************
  2. CDR 2nd Printer Alarm         2. ************************
  3. Check In Hotel/Motel          3. ************************
  4. Check Out Hotel/Motel         4. ************************
  5. Call Forwarding Busy/DA       5. ************************
  6. Clocked Override              6. ************************
  7. Consult/Return                7. ************************
  8. Coverage Callback             8. ************************
  9. Cover Message Retrieve        9. ************************
 10. Data Extension               10. ************************
 11. Time of Day/Date Display     11. ************************
 12. Delete LWC Message           12. ************************
 13. Dial Intercom                13. ************************
 14. Integrated Directory         14. ************************
 15. Directed Call Pickup         15. ************************
 16. Normal/Local Mode Toggle     16. ************************
```

Screen 180. Language Translations — View-buttons (page 3)

```
change display-messages view-buttons                            Page 4 Of 4
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Do Not Disturb                1. ************************
  2. Drop                          2. ************************
  3. Off-Board DS1 Alarm           3. ************************
  4. Manual Exclusion              4. ************************
  5. Do Not Disturb Extension      5. ************************
  6. Flash                         6. ************************
  7. Go To Cover                   7. ************************
  8. Do Not Disturb Group          8. ************************
  9. Group Paging                  9. ************************
 10. Headset On/Off               10. ************************
 11. Hunt Night Service           11. ************************
 12. Coverage Call Identify       12. ************************
 13. Inspect Call Appearance      13. ************************
 14. Internal Auto Answer         14. ************************
 15. Last Number Dialed           15. ************************
 16. Link Failure Alarm           16. ************************
```

Screen 181. Language Translations — View-buttons (page 4)

```
change display-messages view-buttons                            Page 5 Of 5
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Login Security Violation       1. ************************
  2. Cancel Leave Word Call         2. ************************
  3. Lockout Leave Word Call        3. ************************
  4. Leave Word Call Message        4. ************************
  5. Major Hardware Alarm           5. ************************
  6. Manual Message Waiting         6. ************************
  7. Manual Override                7. ************************
  8. ACD Manual-In                  8. ************************
  9. MCT Call Trace Activate        9. ************************
 10. MCT Call Trace Control        10. ************************
 11. Major/Minor Alarm             11. ************************
 12. Message Retrieve              12. ************************
 13. Message Waiting On            13. ************************
 14. Message Waiting Off           14. ************************
 15. Next                          15. ************************
 16. Night Service Activate        16. ************************
```

Screen 182. Language Translations — View-buttons (page 5)

```
change display-messages view-buttons                            Page 6 Of 6
                              LANGUAGE TRANSLATIONS

 English                          Translation

  1. Redirect On No Answer          1. ************************
  2. Normal Display Mode            2. ************************
  3. Personal CO Line               3. ************************
  4. Property Management Fail       4. ************************
  5. Wakeup Printer Alarm           5. ************************
  6. Print Messages                 6. ************************
  7. Priority Calling               7. ************************
  8. PMS Printer Alarm              8. ************************
  9. System Printer Alarm           9. ************************
 10. Number of Queued Calls        10. ************************
 11. Oldest Queued Time            11. ************************
 12. ACD Release                   12. ************************
 13. Ringer Cut-Off                13. ************************
 14. System Reset Alert            14. ************************
 15. Remote Access Violation       15. ************************
 16. Scroll Mode                   16. ************************
```

Screen 183. Language Translations — View-buttons (page 6)

```
change display-messages view-buttons                            Page 7 0f 7
                              LANGUAGE TRANSLATIONS

 English                            Translation

  1. Send All Calls                  1. ************************
  2. TEG Send All Calls              2. ************************
  3. Service Observing               3. ************************
  4. Manual Signalling               4. ************************
  5. Station Security Calal          5. ************************
  6. Stored Number Display           6. ************************
  7. Stroke Counts                   7. ************************
  8. Term Extension Group            8. ************************
  9. Timer                           9. ************************
 10. Facility Test Call Alarm       10. ************************
 11. Trunk ID                       11. ************************
 12. Trunk Name                     12. ************************
 13. Trunk Night Service            13. ************************
 14. UUI Info                       14. ************************
 15. Busy Verification              15. ************************
 16. VDN of Origin Announce         16. ************************
```

Screen 184. Language Translations — View-buttons (page 7)

```
change display-messages view-buttons                            Page 8 0f 8
                              LANGUAGE TRANSLATIONS

 English                            Translation

  1. Vu-Stats Displays               1. ************************
  2. Activate Whisper Page           2. ************************
  3. Whisper Page Answerback         3. ************************
  4. Whisper Page Off                4. ************************
  5. Call Work Code                  5. ************************
  6. Unassigned Button               6. ************************
  7. View Button                     7. ************************
  8. Call Timer                      8. ************************
  9. Add Busy Indicator              9. ************************
 10. Remove Busy Indicator          10. ************************
 11. VIP Wakeup                     11. ************************
 12. VIP Retry                      12. ************************
 13. Crisis Alert                   13. ************************
 14. DID View                       14. ************************
 15. DID Remove                     15. ************************
 16. VIP Check In                   16. ************************
```

Screen 185. Language Translations — View-buttons (page 8)

```
change display-messages view-buttons                          Page   9 of   9
                               LANGUAGE TRANSLATIONS

 English                              Translation

  1.   Station Lock                     1. ************************
  2.   License Error                    2. ************************
  3.   Conference Display               3. ************************
  4.   Conf/Trans Toggle-Swap           4. ************************
  5.   No Hold Conference               5. ************************
  6.   Far End Mute                     6. ************************
```

Screen 186. Language Translations — View-buttons (page 9)

```
change display-messages softkey-labels                     Page   1 of   1
                               LANGUAGE TRANSLATIONS

English      Translation   English      Translation   English      Translation

 1. Acct      1. *****     17. Drop     17. *****     33. SFunc    33. *****
 2.  AD       2. *****     18. Excl     18. *****     34. Spres    34. *****
 3. AdBsy     3. *****     19. GrpPg    19. *****     35. Stats    35. *****
 4. Admin     4. *****     20. HFAns    20. *****     36. Stop     36. *****
 5. AutCB     5. *****     21. IAuto    21. *****     37. Swap     37. *****
 6. BtnVu     6. *****     22. IDial    22. *****     38. Timer    38. *****
 7. CFrwd     7. *****     23. Inspt    23. *****     39. TmDay    39. *****
 8. CnfDs     8. *****     24. Last     24. *****     40. View     40. *****
 9. CnLWC     9. *****     25.  LWC     25. *****     41. Wait     41. *****
10. Cnslt    10. *****     26. Mark     26. *****     42. WspAn    42. *****
11. Count    11. *****     27. Pause    27. *****     43. WspPg    43. *****
12. CPark    12. *****     28. PCall    28. *****
13. CPkUp    13. *****     29. Prog     29. *****
14. CTime    14. *****     30. RmBsy    30. *****
15.  Dir     15. *****     31. RngOf    31. *****
16. DPkUp    16. *****     32.  SAC     32. *****
```

Screen 187. Language Translations — Softkey-Labels

In order to provide unique labeling for abbreviated dialing button types for softkey-labels, the switch replaces the last two characters with digits for the 12-key 8400 and 15-key 8434D phones.

On the softkey label language translation screen, the digits following the “AD” are derived from the button position. If the first button is an AD button, then it is AD1 and the fifteenth button is AD15. All the AD buttons between 1 and 15 have the position number appended to “AD.”

```
change display-messages time-of-day-routing                  Page  1 of  1
                              LANGUAGE TRANSLATIONS

   1.     English: ENTER ACTIVATION ROUTE PLAN, DAY & TIME
      Translation: ****************************************

   2.     English: ENTER DEACTIVATION DAY AND TIME
      Translation: ****************************************

   3.     English:    OLD ROUTE PLAN:      ENTER NEW PLAN:
      Translation: *****************:   *****************:

   4.     English:    OLD ROUTE PLAN:         NEW PLAN:
      Translation: *****************:   **************:

   5.     English: ROUTE PLAN:    FOR       ACT-TIME:
      Translation: **********:   ****     **********:

   6.     English: ROUTE PLAN:    FOR     DEACT-TIME:
      Translation: **********:   ****     **********:
```

Screen 188. Language Translations — Time-Of-Day-Routing

```
change display-messages transfer-conference          Page 1 of 3
                         LANGUAGE TRANSLATIONS

 1.     English: Transfer completed.
    Translation: ***************************************

 2.     English: Call next party.
    Translation: ***********************************************

 3.     English: Press conference to add party.
    Translation: ***********************************************

 4.     English: ~-party conference in progress.
    Translation: ***********************************************

 5.     English: Conference canceled.
    Translation: ***********************************************

 6.     English: Select line ~ to cancel or another line.
    Translation: ***********************************************
```

Screen 189. Language Translations Transfer-conference (page 1)

Message 4

The character " ~ " is a place holder.

English Text	Replacement Info
~-party conference in progress	"~" is replaced with the number of parties currently on the conference call.

Message 6

The character " ~ " is a place holder.

English Text	Replacement Info
Select line ~ to cancel or another line.	" ~ " is replaced with the letter of the line that is on soft hold.

```
change display-messages transfer-conference           Page 2 of 3
                         LANGUAGE TRANSLATIONS

 7.     English: Dial number.
    Translation: ***********************************************

 8.     English: Press transfer to complete.
    Translation: ***********************************************

 9.     English: Hang-up to complete transfer.
    Translation: ***********************************************

10.     English: Dial number or select held party.
    Translation: ***********************************************

11.     English: Select held party to conference.
    Translation: ***********************************************

12.     English: Select line ~ to add party.
    Translation: ***********************************************

13.     English: Select alerting line to answer call.
    Translation: ***********************************************
```

Screen 190. Language Translations Transfer-conference (page 2)

```
change display-messages transfer-conference           Page 3 of 3
                         LANGUAGE TRANSLATIONS

14.     English: Transfer canceled.
    Translation: ***********************************************

15.     English: Connecting to -.
    Translation: ***********************************************

16.     English: Called party - is busy.
    Translation: ***********************************************
```

Screen 191. Language Translations Transfer-conference (page 3)

```
change display-messages transfer-conference          Page 4 of 4
                         LANGUAGE TRANSLATIONS

17.      English: Invalid number dialed -
    Translation: ***************************

18.      English: Party - is not available.
    Translation: ***************************

19.      English: Mute.
    Translation: ****
```

Screen 192. Language Translations Transfer-conference (page 4)

Message 12

The character “ ~ “ is a place holder.

English Text	Replacement Info
Select line ~ to add party.	“ ~ ” is replaced with the letter of the line that is on soft hold.

```
change display-messages vustats                                  Page  1 of  1
                              LANGUAGE TRANSLATIONS

        English                          Translations

   1.   FORMAT                           1.   *******
   2.   NOT DEFINED                      2.   *******
   3.   DOES NOT ALLOW OR REQUIRE ID     3.   *******

   4.   AGENT                            4.   *****************************
   5.   SPLIT/SKILL                      5.   *****************************
   6.   TRUNK GROUP                      6.   *****************************
   7.   VND                              7.   *****************************

   8.   NOT ADMINISTERED                 8.   *****************
   9.   NOT MEASURED                     9.   *****************

  10.   AGENT NOT LOGGED IN             10.   ***********************************
```

Screen 193. Language Translations — VuStats

Listed Directory Numbers

Allows Direct Inward Dialing (DID) numbers to be treated as public Listed Directory Numbers (LDNs). When one of these numbers is direct inward dialed, the calling party is routed to the attendant. The attendant display indicates a Listed Directory Number call and the name associated with the dialed extension.

The number of Listed Directory Numbers that can be assigned varies depending on system configuration. See the *Avaya MultiVantage™ Hardware Solutions Guide* for maximum values.

```
change listed-directory-number                                   Page   1 of   2
                         LISTED DIRECTORY NUMBERS

                          Night Destination:

         Ext     Name                         TN
      1:
      2:                                     1
      3:                                     1
      4:                                     1
      5:                                     1
      6:                                     1
      7:                                     1
      8:                                     1
```

Screen 194. Listed Directory Numbers

Night Destination

Enter the valid assigned extension number that will receive calls to these numbers when Night Service is active.

Valid entries	Usage
0 to **9** (1 to 7 digits)	Enter a night service extension, a recorded announcement extension, a Vector Directory Number, an individual attendant extension, or a hunt group extension.

Ext

Valid entries	Usage
1 to **7** digits	Enter the extension number.

Name

Valid entries	Usage
Up to 27 alphanumeric characters	Enter a name used to identify the Listed Directory Number

TN

Valid entries	Usage
1 to **20**	Enter the Tenant Partition number.

List Usage Report for node-name

```
list usage node-name inde-e72

                                LIST USAGE REPORT
Used By
Processor Channel   Channel Number 4   Destination Node
TFTP Server                            Local Node
```

Screen 195. List Usage Report for local node

The node name, inde-e72, displays as Local Node if it was defined as the local node name on the TFTP Server screen.

TFTP Server

Displays when the node name is used on the TFTP Server screen as the local or server node name.

Local Node Name

Displays when the node name is used on the TFTP Server screen as the local node name.

```
list usage node-name inde-e72

                                LIST USAGE REPORT
Used By
Processor Channel   Channel Number 4   Destination Node
TFTP Server                            Server Node Name
```

Screen 196. List Usage Report for server node name

The node name, inde-e72, displays as Server Node Name if it was defined as the server node name on the TFTP Server screen.

TFTP Server

Displays when the node name is used on the TFTP Server screen as the local or server node name.

Server Node Name

Displays when the node name is used on the TFTP Server screen as the TFTP server node name.

List Usage Report for ip-address

```
list usage ip-address 135.9.76.72

                                LIST USAGE REPORT
Used By
IP Route            Route Number   2                Destination Node
Node Name           IP Name            inde-372    IP Address
Processor Channel   Channel Number 4                Destination Node
TFTP Server                                         Local Node
```

Screen 197. List Usage Report for local node

The node name, inde-e72, displays as Local Node if it was defined as the local node name on the TFTP Server screen.

TFTP Server

Displays when the IP address is used on the TFTP Server screen as the local or server node name.

Local Node Name

Displays when the IP address is used on the TFTP Server screen as the local node.

```
list usage ip-address 135.9.76.72

                                LIST USAGE REPORT
Used By
IP Route            Route Number   2          Destination Node
Node Name           IP Name        inde-372   IP Address
Processor Channel   Channel Number 4          Destination Node
IP Service          Service Type   SAT        Local Node
TFTP Server                                   Server Node Name
```

Screen 198. List Usage Report for server node name

The node name, inde-e72, displays as Server Node Name if it was defined as the TFTP server node name on the TFTP Server screen.

TFTP Server

Displays when the IP address is used on the TFTP Server screen as the local or server node name.

Server Node Name

Displays when the IP address is used on the TFTP Server screen as the server node name.

Locations

Use the Locations screen to provide daylight savings time displays to users, to set the area code for each location, and to administer different location information for multiple switches. You can administer up to 44 location specifications depending on the configuration of your switch if the Multiple Locations field is **y** on the System Parameters Customer Options screen. Otherwise, information for location 1 applies to all your switches.

```
change locations                                                   Page 1 of 3
                                   LOCATIONS

                  ARS Prefix 1 Required for 10-Digit NANP Calls? _

 Number   Name                     Timezone   Daylight-Savings         Number Plan
                                   Offset     Rule                     Area Code

    1     ____________________     _ __:__    __                       ___
    2     ____________________     _ __:__    __                       ___
    3     ____________________     _ __:__    __                       ___
    4     ____________________     _ __:__    __                       ___
    5     ____________________     _ __:__    __                       ___
    6     ____________________     _ __:__    __                       ___
    7     ____________________     _ __:__    __                       ___
    8     ____________________     _ __:__    __                       ___
    9     ____________________     _ __:__    __                       ___
   10     ____________________     _ __:__    __                       ___
   11     ____________________     _ __:__    __                       ___
   12     ____________________     _ __:__    __                       ___
   13     ____________________     _ __:__    __                       ___
   14     ____________________     _ __:__    __                       ___
```

Screen 199. Locations

ARS Prefix 1 Required for 10-Digit NANP Calls?

Valid entries	Usage
y/n	Enter **y** when a 1 must be dialed before all 10-digit NANP calls.

Number

Displays the location number (**1** to **44**). Corresponding entries in each row define the numbering plan and daylight savings rule for each location number.

Name

Valid entries	Usage
up to 15 alphanumeric characters	Identifies the switch associated with each location number.

Timezone Offest

Timezone offset is actually 3 fields (+/-, hour, and minute) that specify how much time each location differs from the system time. This field must be completed for each administered location. Use +00:00 for the time zone offset for a single location switch.

Valid entries	Usage
+	Shows that the time set on this location is a certain amount of time ahead of the system time.
-	Shows that the time set on this location is a certain amount of time behind the system time.

Valid entries	Usage
0 to 23	Shows the number of hours difference between this location and system time.

Valid entries	Usage
0 to 59	Shows the number of minutes difference between this location and system time.

Daylight-Savings Rule

This field must be filled in for each administered location.

Valid entries	Usage
0	No Daylight Savings
1 to 15	Specifies the number for each Daylight-Savings Rule (set up on the Daylight Savings Rule screen) that is applied to this location.

Number Plan Area Code

Valid entries	Usage
0 to 9	Enter the 3-digit numbering plan area code for each location.

Login Administration

When your Avaya MultiVantage is delivered, one customer super-user login and password combination is already defined. You must administer additional logins and passwords, if needed. If you are the super-user, you have full customer permissions and can customize any login you create. The maximum number of customer logins is 11.

As super-user, you can establish permissions for the other logins in your system. You can block access to any object that may compromise switch security. Once the user login is established, set permissions using the Command Permission Categories screen.

Field descriptions for page 1

```
add login                                                          Page 1 of 2
                              LOGIN ADMINISTRATION

             Password of Login Making Change:

         LOGIN BEING ADMINISTERED
                            Login’s Name:xxxxxxx
                              Login Type:
                           Service Level:
   Disable Following a Security Violation?
                                                       Access to INADS Port? _

        LOGIN’S PASSWORD INFORMATION
                        Login’s Password:
                Reenter Login’s Password:
      Password Aging Cycle Length (Days):

      LOGOFF NOTIFICATION
        Facility Test Call Notification?     Acknowledgment Required?
             Remote Access Notification?     Acknowledgment Required?

ACCESS SECURITY GATEWAY PARAMETERS
Access Security Gateway?
```

Screen 200. Login Administration

Password of Login Making Change

You can make changes to any login with permissions less than your own. You must enter your password to save any changes you make to this screen.

Valid entries	Usage
User password	Enter your password.

Login's Name

This display-only field shows the login name specified with the add command.

Login Type

Valid entries	Usage
Customer	Enter **customer** to indicate the login belongs to a customer. This is the only valid value for customer logins.

Service Level

Valid entries	Usage
super-user	Enter super-user to indicate the user is a super-user. Super-user logins can use the add, change, display, list, and remove commands for all customer logins and passwords.
non-super-user	Enter non-super-user to indicate the user is a non-super-user. Non-super-user logins can change their own password, but are otherwise limited to permissions set by super-user.

Disable Following a Security Violation

This field only appears when the SVN Login Violation Notification field is set to y on the Security-Related System-Parameters screen.

Valid entries	Usage
y/n	Enter **y** to have the login disabled following a security violation.

Access to INADS Port

This field only appears when the Customer Access to INADS Port field is set to y on the Maintenance-Related System-Parameters screen.

Valid entries	Usage
y/n	Enter **y** to allow a user to access the remote administration port.

Login's Password

Passwords must be 4 to 11 characters in length and contain at least 1 alphabetic and 1 numeric character. Passwords do not display as you type.

Valid entries	Usage
A-Z, a-z, 0-9, and ! & * ? ; ' ^ ( ) , . : -	Enter the initial password for login, assigned by the super-user.

Re-enter Login's Password

Passwords do not display as you type.

Valid entries	Usage
A-Z, a-z, 0-9, and ! & * ? ; ' ^ () , . : -	Re-enter the initial password for the login, assigned by the super-user.

Password Aging Cycle Length (Days)

For security, you should assign password aging to all passwords. The system tracks a password from the day the login was created, or the day the user last changed the password.

Valid entries	Usage
1 - 99	Enter the number of days the password remains valid.

Facility Test Call Notification

For security, this field should be set to y for all logins.

Valid entries	Usage
y/n	Enter **y** to have the user receive notification at logoff if Facility Test Notification is still administered.

Facility Test Call Notification — Acknowledgment Required

Valid entries	Usage
y/n	Enter **y** to require the user to acknowledge they want to logoff while Facility Test Notification is still administered.

Remote Access Notification

For security, this field should be set to y for all logins.

Valid entries	Usage
y/n	Enter **y** to have the user receive notification at logoff if remote access is still administered.

Remote Access Notification — Acknowledgment Required

Valid entries	Usage
y/n	Enter **y** to require the user to acknowledge they want to logoff while remote access is still administered.

Access Security Gateway

This field appears only if the Access Security Gateway (ASG) field on the System-Parameters Customer-Options screen is set to y.

Valid entries	Usage
y/n	Enter **y** to have ASG administered for the login ID.

Field descriptions for page 2

This page only appears if the Access Security Gateway field is set to **y** on page 1.

```
add login                                                     Page 2 of 2
                     ACCESS SECURITY GATEWAY LOGIN ADMINISTRATION

                    Blocked?: n
System Generated Secret Key?: n       Secret Key: ____________________



EXPIRATION CRITERIA
   Expiration Date: __/__/____
Number of Sessions: __

RESTRICTION CRITERIA
Restrict Days of Week
   Monday? n     Tuesday? n     Wednesday? n     Thursday? n     Friday? n
 Saturday? n      Sunday? n

 Restrict From Time: __:__      Restrict To Time: __:__
```

Screen 201. Access Security Gateway Login Administration

Blocked

Setting the Blocked field to **y** does not remove the login ID from the system.

Valid entries	Usage
y/n	Enter **y** to temporarily disable the login ID from accessing the system through the ASG interface.

System Generated Secret Key

Valid entries	Usage
y/n	Enter **y** to generate the secret key by the system and display it to the administrator.

Secret Key

The value of this field is only displayed during the initial add or change login command to allow it to be programmed into any ASG Site Manager, ASG Mobile, or ASG Pass-Key used to interact with ASG for user authentication.

If the System Generated Secret Key field is set to **y**, the system generates the secret key. If the System Generated Secret Key field is set to n, the administrator is required to enter a 20-digit octal value. It is important to note the value of the secret key so you can enter it into response generation devices.

Valid entries	Usage
20-digit octal string using 0-7 as possible digits	Enter the 20-digit octal value used by both the lock and the key to verify user authenticity. The last digit must be 0 and the second to last digit must be either 0, 2, 4, or 6.

Expiration Date

Expiration of the login ID occurs at 23:59:59 of the entered date. This field consists of three fields separated by forward slashes (for example, 01/01/1999). Expiration of a login ID does not remove the login ID from the system. The system accepts administration of year 2000 dates.

If an login ID requiring ASG authentication has expired, the login ID for the standard login is not available. If an login ID not requiring ASG login has expired, the login ID still may be active for the standard login as long as expiration criteria associated with that component of the login has not been satisfied.

Valid entries	Usage
mm = month, dd = day, and yyyy = year (later than the current date)	Enter the date when the associated login ID expires.

Number of Sessions

Expiration of a login ID administered with both Expiration Date and Number of Sessions expiration criteria is based on whichever criteria is satisfied first.

Valid entries	Usage
1 to **999**	Enter the number of times the login ID can be used to access the system via the ASG interface.

Restrict Days of Week

This field consists of seven subfields that correspond to the seven days of the week. Each subfield specifies whether the corresponding login ID is restricted from accessing the system via ASG on the day indicated. Access restrictions imposed by this field apply to the entire day unless limited by time restrictions.

Valid entries	Usage
y/n	Enter **y** to restrict the login from accessing the system on the associated day of the week.

Restrict From Time and Restrict to Time

These two fields are separated by colons, for example, hh:mm where hh = hour and mm = minute. Periods that span an interval that crosses a day boundary are specified by setting the Restrict From Time field greater than the Restrict To Time field. For example, a Restriction From Time of 17:00 and a Restrict To Time of 08:00 limits access to the traditional working hours, 8 to 5.

When used with the Restrict Days of Week field, overnight periods start on the days where access is not restricted by the Restrict Day of Week field.

If the Restrict Days of Week field is not specified, the restricted time interval specified by the Restrict From Time to Restrict To Time fields applies to every day of the week.

Valid entries	Usage
00:00 to **23:59**	Enter the time interval each day the login ID is blocked from accessing the system via the ASG interface.

Loudspeaker Paging

The Loudspeaker Paging screen administers voice paging, deluxe voice paging, and chime paging.

```
change paging loudspeaker                                      Page   1 of   1
                             LOUDSPEAKER PAGING

                                   CDR? _
            Voice Paging Timeout (sec): ___
           Code Calling Playing Cycles: _

PAGING PORT ASSIGNMENTS
                 Voice Paging      Code Calling
 Zone  Port      TAC  COR  TN      TAC  COR  TN     Location:
   1: _______    ____ __   _       ____ __   _      ___________________________
   2: _______    ____ __   _       ____ __   _      ___________________________
   3: _______    ____ __   _       ____ __   _      ___________________________
   4: _______    ____ __   _       ____ __   _      ___________________________
   5: _______    ____ __   _       ____ __   _      ___________________________
   6: _______    ____ __   _       ____ __   _      ___________________________
   7: _______    ____ __   _       ____ __   _      ___________________________
   8: _______    ____ __   _       ____ __   _      ___________________________
   9: _______    ____ __   _       ____ __   _      ___________________________
 ALL: _______    ____ __   _       ____ __   _
```

Screen 202. Loudspeaker Paging

CDR

This field determines whether CDR data is collected for the paging ports.

Valid entries	Usage
y/n	Enter **y** if you want the switch to collect CDR data on the paging ports.

Voice Paging Timeout (sec)

This field limits the duration of voice pages. When this interval ends, calls are disconnected. To determine the best setting, time the typical pages you expect to broadcast and then add another 4 to 5 seconds.

Valid entries	Usage
10 to **600** seconds.	Enter the maximum number of seconds you want any page to last.
blank	The field may not be blank when you administer voice paging.

Code Calling Playing Cycles

This field sets the number of times a chime code will play when a user makes a chime page. To determine the best setting, consider who your code calling users are and whether they are likely to hear the code chime the first time.

Valid entries	Usage
1 to **3**	Enter the number of times you want the chime code to play when a user makes a page.
blank	The field may not be blank when you administer chime paging (code calling).

Port

This field assigns a port on an auxiliary trunk circuit pack to a paging zone.

Valid entries	Usage
a port ID	Assign an available port to each zone you want to use. You cannot assign the same port to different zones.
blank	Leave this field blank for unused paging zones.

NOTE:
To use a port that has no hardware associated with it, place an ’X’ in this field.

Voice Paging — TAC

This field assigns a Trunk Access Code (TAC) to a paging zone. Users dial this code to make a page to a specific zone. One TAC must be assigned to each zone you want to use. Two zones cannot have the same TAC. If you enter a TAC in the ALL field, users may activate speakers in all the zones by dialing that code.

Valid entries	Usage
1- to 4-digits	Enter a Trunk Access Code (TAC) allowed by your dial plan.
*****	May be used as first digit.
#	May be used as first digit.
blank	Leave this field blank for unused paging zones.

Voice Paging — COR

This field assigns a Class of Restriction to a paging zone.

Valid entries	Usage
0 through **95**.	You can assign different classes of restriction to different zones.
blank	Leave this field blank for unused paging zones.

Voice Paging — TN

Valid entries	Usage
1 to **20** (DEFINITY Server SI / DEFINITY Server CSI)	If your system uses Tenant Partitioning, you may use this field to assign a paging zone to a specific tenant partition.
1 to **100** (DEFINITY Server R)	

Code Calling — TAC

This field assigns a Trunk Access Code (TAC) to a paging zone. Users dial this code to make a page to a specific zone. One TAC must be assigned to each zone you want to use. Two zones cannot have the same TAC. If you enter a TAC in the ALL field, users may activate speakers in all the zones by dialing that code.

Valid entries	Usage
1- to 4-digits	Enter a Trunk Access Code (TAC) allowed by your dial plan.
*****	May be used as first digit.
#	May be used as first digit.
blank	Leave this field blank for unused paging zones.

Code Calling — COR

This field assigns a Class of Restriction to a paging zone.

Valid entries	Usage
0 through **95**.	You can assign different classes of restriction to different zones.
blank	Leave this field blank for unused paging zones.

Code Calling — TN

Valid entries	Usage
1 to **20** (DEFINITY Server SI / DEFINITY Server CSI)	If your system uses Tenant Partitioning, you may use this field to assign a paging zone to a specific tenant partition.
1 to **100** (DEFINITY Server R)	

Location

Valid entries	Usage
1 to 27 characters	Assign a descriptive name for the physical location corresponding to each zone. Typical entries might be “conference room A,” “warehouse,” or “storeroom.”

Related topics

Refer to ‘‘Setting up voice paging over loudspeakers’’ on page 463 or ‘‘Setting up chime paging over loudspeakers’’ on page 466 for instructions.

Refer to ‘‘Loudspeaker paging’’ on page 1633 for a description of the feature.

Media-Gateway

This screen administers the media gateways and their corresponding ports. It is required to exist on all Communications Controllers, both external and internal, that support Media Gateways.

Field descriptions for page 1

```
add media-gateway x                                              Page   1 of 1
                                MEDIA-GATEWAY

        Number: ___
          Name: ____________________ Identifier: _____________________
    IP Address: _______________     MAC Address: _________________
Network Region: ____                   Location: __
     Site Data: ____________________ Registered? _
                ____________________

                  Slot Module Type
                  V1: # ________
                  V2: __________
                  V3: __________
                  V4: __________



                  V8: messaging-analog
                  V9: gateway-accountements
```

Screen 203. Media-Gateway Administration screen

Number

A display-only field showing the assigned media-gateway number.

Name

This field is for customer naming of the gateway.

Valid entries	Usage
1- to 20-alphanumeric character string or blank	Enter the alpha-numeric characters for identification.

Identifier

This field is for the controller to identify the gateway.

Valid entries	Usage
1 to 20 characters or blank	Enter the serial number or other alpha-numeric characters for identification.

IP Address

A display-only field containing the IP address of the media gateway.

MAC Address

A display-only field containing the MAC address of the media gateway.

Network Region

This field is used by the communications controller to allocate resources from the nearest media gateway.

Valid entries	Usage
2 to 3 characters or blank	Enter a number between 1-X or blank where X is the maximum value for the different communications controllers.

Location

This field refers to a time-zone offset, daylight savings rule and number plan area code.

Valid entries	Usage
2 characters	Enter a number between 1-X or blank where X is the maximum value for the different communications controllers.

Site Data

This field is for general site information.

Valid entries	Usage
2 lines of 20 characters or blank	Enter the alpha-numeric characters for identification.

Registered

A display-only field indicating whether a Media Gateway is registered with a communications controller.

Valid entries	Usage
1 character	Enter a number between 1-X or blank where X is the maximum value for the different communications controllers.

Module Type

Fields V1 through V4 list the Media Gateway media modules.

Valid entries	Usage
analog	Enter the actual hardware media module.
bri	
dcp	
icc	
voip	
ds1	

Field V8 lists a “virtual” media modules (available in August).

Valid entries	Usage
messaging-analog or blank	Messaging will be administered in this slot.

Field V9 lists a “virtual” media modules (available in December).

Valid entries	Usage
gateway-announcements or blank	Announcement will be administered in this slot.

Meet-me Vector Directory Numbers

Field descriptions for page 1

```
LIST MEET-ME-VDN

                   MEET-ME VECTOR DIRECTORY NUMBER

                                  Access          Vec Control
Name                         Ext  Code   COR  TN  Num   Ext

Secure Meet-me Conference   4000   *      1    1   1
Nonsecure Meet-me Conference 4006         1    1   2    84590
```

Screen 204. Meet-me Vector Directory Numbers screen

If the Access Code field shows an asterisk (*), an access code is assigned. If the Access Code field is blank, no access code is assigned. The access code displays for administrators with super-user permissions (such as *init*).

Mode Code Related System Parameters

This screen establishes parameters associated with the Mode Code Voice Mail System Interface.

NOTE:
You can only administer this screen if the Mode Code Interface field on the Feature-Related System Parameters screen is set to **y**.

Field descriptions for page 1

```
                                                                 Page   1
                      MODE CODE RELATED SYSTEM PARAMETERS

MODE CODES  (FROM SWITCH TO VMS)
      Direct Inside Access: __
Direct Dial Access - Trunk: __
         Internal Coverage: __
         External Coverage: __

           Refresh MW Lamp: __

     System In Day Service: __
   System In Night Service: __

                     OTHER RELATED PARAMETERS
DTMF Duration On (msec): __ Off (msec): __ Sending Delay (msec):__
 VMS Hunt Group Extension : ____
Remote VMS Extensions - First:     Second:
```

Screen 205. Mode Code Related System Parameters screen

Direct Inside Access

This value defines a mode code that the switch sends when a caller at an internal extension dials the Voice Mail System (VMS) access number.

Valid entries	Usage
0 to **9, #, *, #00**	Up to six digits that may include these characters

Direct Dial Access - Trunk

This value defines a mode code that the switch sends when an external caller dials the VMS access number.

Valid entries	Usage
0 to **9, #, *, #00**	Up to six digits that may include these characters

Internal Coverage

This value defines a mode code that the switch sends when an internal caller tries to reach a user at another extension and the call goes to the user's voice mail coverage.

Valid entries	Usage
0 to **9**, **#**, *****, **#00**	Up to six digits that may include these characters

External Coverage

This value defines a mode code that the switch sends when an external caller tries to reach a user at another extension and the call goes to the user's voice mail coverage.

Valid entries	Usage
0 to **9**, **#**, *****, **#00**	Up to six digits that may include these characters

Refresh MW Lamp

This value defines a mode code that the switch sends during a system level 3 or higher reset that requests the VMS to refresh the Message Waiting (MW) lamps.

Valid entries	Usage
0 to **9**, **#**, *****, **#00**	Up to six digits that may include these characters

System In Day Service

This value indicates to the VMS that the Avaya MultiVantage has changed from Night to Day Service.

Valid entries	Usage
0 to **9**, **#**, *****, #**11**	Up to six digits that may include these characters

System In Night Service

This value indicates to the VMS that the Avaya MultiVantage has changed from Day to Night Service.

Valid entries	Usage
0 to **9**, **#**, *****, #**12**	Up to six digits that may include these characters

Other Related Parameters

DTMF DURATION ON

Valid entries	Usage
Between **75** and **500** in multiples of 25	Define in milliseconds the length of mode code digits sent to the VMS. This field cannot be blank

OFF

Valid entries	Usage
Between **75** and **200** in multiples of 25	Define in milliseconds the pause between mode code digits as they are sent to the VMS. This field cannot be blank.

Sending Delay

Valid entries	Usage
75 to **1000** in multiples of 25	Define in milliseconds the delay between the time the switch receives answer supervision from the VMS and the time the first mode code digit is sent. This field cannot be blank.

VMS Hunt Group Extension

A check is made to verify that a valid hunt group extension is entered, but a check is not made to verify that the hunt group members are VMI extensions.

Valid entries	Usage
Extension of a hunt group containing VMI extensions.	

Remote VMS Extensions - First

You can administer this field if the Mode Code for Centralized Voice Mail field on the System-Parameters Customer-Options screen is set to **y**. Specifies the first remote UDP VMS hunt group extension.

Valid entries	Usage
Remote assigned hunt group extension	Enter the first UDP VMS hunt group extension.

Remote VMS Extensions - Second

You can administer this field if the Mode Code for Centralized Voice Mail field on the System-Parameters Customer-Options screen is set to **y**. Specifies the second remote UDP VMS hunt group extension.

Valid entries	Usage
Remote assigned hunt group extension	Enter the second UDP VMS hunt group extension. This extension cannot be the same as the first Remote VMS Extension.

Modem Pool Group

There are two types of conversion resources for Modem Pooling. The first type, an *integrated conversion resource*, is a circuit pack that emulates a Trunk Data Module connected to a 212A-type modem. Two conversion resources are on each circuit pack.

The second type, a *combined conversion resource*, is a separate Trunk Data Module and modem administered as a unit. The Trunk Data Module component of the conversion resource may be either a Modular Trunk Data Module (MTDM) or 7400A Data Module and connects to a digital port using Digital Communications Protocol (DCP); the modem connects to an analog port.

Field descriptions for page 1

```
change modem-pool num                                                   Page 1 of 1
                                   MODEM POOL GROUP
                    Group Number: 1                    Group Type: integrated
Receiver Responds to Remote Loop? n               Hold Time (min): 5
           Send Space Disconnect? y     Receive Space Disconnect? y
                    CF-CB Common? y   Loss of Carrier Disconnect? y

Speed: LOW/300/1200      Duplex: full     Synchronization: a/sync

CIRCUIT PACK ASSIGNMENTS
               Circuit Pack   Circuit Pack
               Location       Location
               1: ___          9: ___
               2: ___         10: ___
               3: ___         11: ___
               4: ___         12: ___
               5: ___         13: ___
               6: ___         14: ___
               7: ___         15: ___
               8: ___         16: ___
```

Screen 206. Modem Pool Group — Integrated screen if Group Type is integrated

```
change modem-pool num                                             Page 1 of 1
                          MODEM POOL GROUP
               Group Number: _          Group Type: combined
                 Modem Name: ______ Hold Time (min): 5_
                 Time Delay: 0_          Direction: two-way
Answer Supervision Timeout(sec): _

    Speed: LOW/300/1200___    Duplex: full    Synchronization: async

PORT PAIR ASSIGNMENTS
      Analog Digital    Analog Digital     Analog Digital      Analog Digital
    1: _____ _____    9: _____ _____    17: _____ ______    25: _____ _____
    2: _____ _____   10: _____ _____    18: _____ ______    26: _____ _____
    3: _____ _____   11: _____ _____    19: _____ ______    27: _____ _____
    4: _____ _____   12: _____ _____    20: _____ ______    28: _____ _____
    5: _____ _____   13: _____ _____    21: _____ ______    29: _____ _____
    6: _____ _____   14: _____ _____    22: _____ ______    30: _____ _____
    7: _____ _____   15: _____ _____    23: _____ ______    31: _____ _____
    8: _____ _____   16: _____ _____    24: _____ ______    32: _____ _____
```

Screen 207. Modem Pool Group — Combined screen if Group Type is combined

Group Number

A display-only field when the screen is accessed using an administration command such as **add** or **change**.

Group Type

This field designates what physical model pool you are going to.

Valid entries	Usage
integrated	Maps to the Pooled Modem circuit pack.
combined	Maps to an external modem pool (when you have a data module and a modem).

Receiver Responds to Remote Loop

This field appears only when the Group Type field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to allow the far-end modem to put conversion resource into loop back mode.

Modem Name

Indicates the name of the modem pool. This field appears only when the Group Type field is **combined**.

Valid entries	Usage
1- to 6-alphanumeric character string	

Hold Time (min)

Enter the maximum number of minutes that a conversion resource in the group may be held while a call waits in a queue or reserved after Data Call Preindication.

Valid entries	Usage
1 through **99**	

Send Space Disconnect

This field appears only when the Group Type field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to allow the conversion resource to send 4 seconds of space before disconnecting.

Time Delay

Enter the time delay in seconds to insert between sending the ringing to the modem and the off-hook alert to the data module. This field appears only when the Group Type field is **combined**.

Valid entries	Usage
0 through **255**	

Receive Space Disconnect

This field appears only when the Group Type field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to allow the conversion resource to disconnect after receiving 1.6 seconds of space.

CF-CB Common

This field appears only when the Group Type field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to indicate that the CF and CB leads on the conversion resource are logically connected.

Direction

Enter the direction of the call for which modem pool will operate. This field appears only when the Group Type field is **combined**.

Valid entries	Usage
incoming	Converts an analog signal to digital for the data endpoint.
outgoing	Converts analog to digital (or digital to analog) for data calls.
two-way	Allows incoming and outgoing data communication.

The following fields (Speed, Duplex, and Synchronization) cannot be filled out for the “integrated” pooled modem screens but can be assigned on the “combined” pooled modem screen. The integrated conversion resource automatically will adjust its speed and synchronization to the endpoint it is connected to. In synchronous mode, the integrated modem pool can operate at 1200 baud. In asynchronous mode, it can operate at 300 or 1200 baud. Full-duplex operation is always used.

Loss of Carrier Disconnect

This field appears only when the Group Type field is **integrated**.

Valid entries	Usage
y/n	Enter **y** to permit conversion resource to disconnect if it detects a dropped carrier.

Answer Supervision Timeout (sec)

Enter the period of time to wait before the far-end answers. This field appears only when the Group Type field is **combined**.

Valid entries	Usage
1–255	
0	No answer supervision

Speed

Enter the communication speed in bits per second of the conversion resources in the group. Enter one to three speeds separated by slashes (for example, 300/1200/2400) to indicate a maximum of three running speeds.

Valid entries	Usage
LOW	0 to 300 blind sampled
300	
1200	
2400	
4800	
9600	
19200	

Duplex

Indicates the duplex mode of the conversion resources in the group.

Valid entries	Usage
full	Can talk and listen at the same time.
half	Cannot talk and listen at the same time.

Synchronization

Indicates the synchronization mode of the conversion resources in the group.

Valid entries	Usage
sync	Synchronous
async	Asynchronous

CIRCUIT PACK ASSIGNMENTS are optional on “integrated” conversion resource screens only.

Circuit Pack Location

Enter the port associated with the conversion resource on the integrated modem pool circuit pack.

Valid entries	Usage
01 through **22**	First and second characters are the cabinet number
01 through **03**	
01 (G3s)	
A through **E**	Third character is the carrier
01 through **20**	Fourth and fifth characters are the slot number
01 through **32**	Sixth and seventh characters are the circuit number

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

PORT PAIR ASSIGNMENTS are optional on “combined” pooled modem screens only.

Analog Digital

Enter the port numbers of the modem/TDM pair in a conversion resource.

NOTE:
Two port entries are required.

Valid entries	Usage
01 through **22**	First and second characters are the cabinet number
01 through **03**	
01 (G3s)	
A through **E**	Third character is the carrier
01 through **20**	Fourth and fifth characters are the slot number
01 through **32**	Sixth and seventh characters are the circuit number

For example, 01A0612 is in cabinet 01, carrier A, slot 06, and circuit number (port) 12.

Multifrequency-Signaling-Related System Parameters

This screen sets the system parameters associated with multifrequency signaling. Screen 208 on page 992 appears when Incoming Call Type is **group-ii-mfc** and Outgoing Call Type is **none**. Screen 209 on page 1000 appears when both Incoming Call Type and Outgoing Call Type are **group-ii-mfc**.

If the field, Use COR for All Group II Responses, is set to **y**, the Group II Called Party Category and Use COR for Calling Party Category fields do not appear.

```
change system-parameters multifrequency-signaling                Page   1 of   4

             MULTIFREQUENCY-SIGNALING-RELATED SYSTEM PARAMETERS

          Incoming Call Type:                      ANI Prefix:
          Outgoing Call Type:                     ANI for PBX:
       Maintenance Call Type:            NEXT ANI DIGIT
         Test Call Extension:                        Incoming:
      Interdigit Timer (sec):                        Outgoing:
     Maximum Resend Requests: _
   Received Signal Gain (dB): _
Transmitted Signal Gain (dB): -
                          Request Incoming ANI (non-AAR/ARS)?
                   Outgoing Forward Signal Present Timer (sec):
                    Outgoing Forward Signal Absent Timer (sec):
MF Signaling Intercept Treatment - Incoming? _  Outgoing: ____________ _____
                             Collect All Digits Before Seizure?
                  Overlap Sending on Link-to-Link Tandem Calls?
          Private Group II Permissions and Public Interworking?
                             Convert First Digit End-of-ANI To: _
                                Group II Called Party Category:
                            Use COR for Calling Party Category?
                   Outgoing Shuttle Exchange Cycle Timer (sec):
```

Screen 208. Multifrequency-Signaling-Related System Parameters

The ANI Prefix, ANI for PBX, and Collect All Digits Before Seizure fields appear only when the value of the Outgoing Call Type field is **group-ii-mfc** or **mfe**.

If Collect All Digits Before Seizure is **y**, Overlap Sending on Link-to-Link Tandem Calls and Convert First Digit End-of-ANI are not displayed.

Incoming Call Type

This field defines the signal type that a CO uses to place an incoming call to the PBX.

Valid entries	Usage
group-ii-mfc	If the value of this field is **group-ii-mfc**, the second page of the screen will display entries for all group-I, group-II, group-A, and group-B signal types with a set of default values (see page 2 of screen).
non-group-ii-mfc	If the value is **non-group-ii-mfc**, the second page displays only group-I and group-A signal types.
mfe	Use only in Spain (multi-frequency Espanol)

ANI Prefix

This field appears only when Outgoing Call Type is **group-ii-mfc** or **mfe**.

Valid entries	Usage
1 to 6 digits	Enter the prefix to apply to an extension when ANI is sent to the CO.

Outgoing Call Type

This field defines the signal type that the PBX uses to place an outgoing call into a CO.

Valid entries	Usage
group-ii-mfc	If the content of this field is **group-ii-mfc**, the system displays the third page of the screen. The third page displays entries for all group-I, group-II group-A, and group-B signal types with a set of default values.
mfe	Use only in Spain (multi-frequency Espanol)
none	If the content of this field is **none**, the system does not display the third page. In addition, Outgoing Forward Signal Present Timer, Outgoing Forward Signal Absent Timer, ANI Prefix, ANI for PBX, Next ANI Digits, and Collect All Digits Before Seizure will not display on Field descriptions for page 1.

ANI for PBX

This field appears only when Outgoing Call Type is **group-ii-mfc** or **mfe**.

Valid entries	Usage
2 to **15**	Enter the PBX identification number that is sent to the CO when ANI is requested (by the CO) on a particular call but is not available, such as on tandem tie trunk calls.
blank	Use for tandeming. If this field is blank, you must enter a value in the ANI-Not-Available field.

MFE Type

This field only appears when Incoming Call Type is **mfe** and the Outgoing Call Type is **mfe** or **none.**

Valid entries	Usage
2/5	Determines which public signaling the switch will use.
2/6	

Maintenance Call Type

Valid entries	Usage
1	The Belgium maintenance sequence is indicated when the CO sends an MFC maintenance tone.
2	The Saudi Arabian sequence is indicated when the CO sends an MFC maintenance tone.

Next ANI Digit

This field appears when Outgoing Call Type is **group-ii-mfc**.

Valid entries	Usage
next-digit **next_ani_ digit** **send-ani**	Enter a value to determine whether the Next ANI Digit signal will be the same as the “send-ani” signal or the “next-digit” signal or another signal defined as “next_ani_ digit.”

Test Call Extension

Valid entries	Usage
An unassigned extension	Specifies the destination of a call between the CO and the PBX that tests R2-MFC signaling.
1	A test call extension is the destination.

Interdigit Timer (sec)

Specify the maximum number of seconds the switch will wait for the first forward signal (digit) to arrive, and for subsequent digits to arrive. Intercept returns to the calling party if this timer expires.

Valid entries	Usage
1 to **255**	This number must be less than the number of seconds entered in the short interdigit timer.

Maximum Resent Requests

Valid entries	Usage
1 to **99**	Enter the threshold number of resend type MFC signals your switch accepts during an outgoing call.
1	The call is dropped if one resend signal is received.
blank	An unlimited number of resend requests is allowed.

Received Signal Gain (-Loss) (dB)

Valid entries	Usage
-15 to **3**	Enter the number for the loss/gain when the MFC port listens to the trunk port. Your switch listens with a range of -5 to -35 and this value moves the range (for example, a value of -5 provides a range of -10 to -40).

Transmitted Signal Gain (-Loss) (dB)

Valid entries	Usage
-15 to **3**	Enter the number for the loss/gain when the trunk port listens to the MFC port. The MFC port generates at -5 for MFC and -8 for MFE, and this field adds gain or loss to the starting value of -5.

Outgoing Forward Signal Present Timer (sec)

This field appears only when the value of Outgoing Call Type is **group-ii-mfc**.

Valid entries	Usage
1 to **255**	Enter the maximum number of seconds to elapse between signals on a call. This timer runs when MFC tones are being sent or received on an outgoing call. The timer starts (and restarts) when the switch begins sending a forward signal and stops when the switch receives the backward signal.

Outgoing Forward Signal Absent Timer (sec)

This field appears only when the content of Outgoing Call Type is **group-ii-mfc**.

Valid entries	Usage
11 to **255**	Enter the maximum number of seconds to elapse between forward signals on outgoing calls. The timer starts (and restarts) when a forward tone is taken off the link and it stops when the next forward tone is applied to the link.

MF Signaling Intercept Treatment - Incoming

Valid entries	Usage
y	Send the group B signal for the intercept to the CO and play intercept tone on the trunk.
n	Use normal DID/TIE/ISDN intercept treatment.

MF Signaling Intercept Treatment - Outgoing

Valid entries	Usage
announcement	Plays a recorded announcement for outgoing calls that cannot be completed as dialed. You select and record the message. Enter the extension number for the announcement in the associated field.
tone	Plays intercept tone for outgoing calls that cannot be completed as dialed.

Overlap Sending on Link-to-Link Tandem Calls

An Avaya MultiVantage with this field set to **y,** when tandeming calls between switches will send ANI for PBX to the terminating switch if that switch requests ANI before the Avaya MultiVantage receives it from the originating switch. The terminating switch may request ANI before the receipt of the last address digit if it is not an Avaya MultiVantage or it is an Avaya MultiVantage with the Request Call Category at Start of Call field set to **y**.

Valid entries	Usage
y/n	If **y**, Avaya MultiVantage sends and receives digits one digit at a time instead of enbloc. (With enbloc, digits are not sent until the entire group of digits is received).

Private Group II Permissions and Public Interworking

Valid entries	Usage
y/n	If **y**, Avaya MultiVantage: — Sends the category for MFC ANI for the COR of the originating party for non-private-MFC-trunk to MFC-private-trunk calls. — Sends the Group II category received over the incoming private trunk as the outgoing Group II category on tandem private MFC calls. — Applies MFC group II-CPC termination restrictions on incoming MFC private trunk calls. — Checks station permissions if you call forward off-net.

Convert First Digit End-of-ANI To

Valid entries	Usage
0 to **9**	Enter the digit used when the incoming initial end-of-ani or end-of-dial MF signal is converted on a per-switch basis.

Collect All Digits Before Seizure

This field appears only when Outgoing Call Type is **group-ii-mfc** or **mfe**.

Valid entries	Usage
y	The system collects all the digits before seizing the trunk and the ANI Req field on the AAR and ARS Digit Conversion Table does not apply.
n	Enter **n** to control ANI collection via the ARS screens**.**

Request Incoming ANI (non-AAR/ARS)

This field only applies if the incoming call via the R2-MFC trunk is terminating to a local station on this PBX.

Valid entries	Usage
y/n	If **y**, ANI should be requested on incoming R2-MFC calls.

Forward Cycle Timer (sec)

This field appears only if Incoming Call Type is **mfe** and Outgoing Call Type is **mfe** or **none.**

Valid entries	Usage
1 to **255**	Enter the number of seconds to wait to receive the check frequency after sending an MFE signal. The switch drops the call if the time runs out before it receives check frequency.

Backward Cycle Timer (sec)

This field appears only if Incoming Call Type is **mfe** and Outgoing Call Type is **mfe** or **none.**

Valid entries	Usage
1 to **255**	Enter the number of seconds to wait to send the check frequency after receiving an MFE signal.

Incomplete Dial Timer (sec)

This field appears only if Incoming Call Type is **mfe** and Outgoing Call Type is **mfe** or **none.**

Valid entries	Usage
45 to **255**	Enter the number of seconds to wait from the start of a call until the end of the check frequency of the last signal. The switch drops the call if the time runs out before it receives the check frequency.

Outgoing Start Timer (sec)

The field appears only if Incoming Call Type and Outgoing Call Types are both **mfe**.

Valid entries	Usage
1 to **255**	Enter the number of seconds to time from seizure until the beginning of the first Group A signal from the receiving end, and from the end of the check frequency until the beginning receipt of the first digit following the Group II signal.

Group II Called Party Category

Enter the type of group II signals that should be used on the outgoing R2-MFC call

Note: For India, the ANI can be requested without the call category information.

Valid entries	Usage
user-type	The type of phone making the call determines the type of group II signal that the PBX sends (normal = ordinary phone set, attendant = attendant console, data-call = data modules and similar data endpoints).
call-type	The dialed digits determine the type of group II signal that the PBX sends.

Use COR for Calling Party Category

Indicates the category to send with ANI if requested on an outgoing R2-MFC call.

Valid entries	Usage
y	Use the calling facility’s COR to determine category.
n	Use the calling party’s user-type COR to determine category.

Use COR for all Group II Responses

This field only appears if the Outgoing Call Type field is set to **group-ii-mfc**.

Valid entries	Usage
y/n	Enter **y** to allow the COR administered category to be used for both the calling party and called party categories.

Outgoing Shuttle Exchange Cycle Timer (sec)

This field applies only to calls made from Avaya MultiVantage.

Valid entries	Usage
1 to **25**	Enter the number of seconds to time an exchange cycle (starts when the far end requests a digit until Avaya MultiVantage sends the requested digit).

Field descriptions for page 2

The fields on Page 2 define call category and ANI information. For India, the ANI can be requested without the call category information.

```
                                                        Page  2 of 3
          MULTIFREQUENCY-SIGNALING-RELATED SYSTEM PARAMETERS


    Request Call Category at Start of Call: n
          Restart ANI from Caller Category? y
             Number of Incoming ANI Digits: 0
             Number of Outgoing ANI Digits: 0
            Truncate station number in ANI: no
  ANI Source for Forwarded & Covered Calls: _________
Address Digits Include End-of-Digits Signal? n
        Call Category for Vector ii-digits? n
              Request CPN at Start of Call? y
         Do Not Send Group B Signals to CO? y

                            INCOMING     OUTGOING

        ANI Available:          ___          ___
    ANI Not Available:          ___          ___
```

Screen 209. Multifrequency-Signaling-Related Parameters screen

Request Call Category at Start of Call

Indicates that the Send-ANI backward signal requesting for the caller-category information will be sequenced differently in the MFC signaling flow. The Caller-category Request backward signal is disjointed from the ANI request.

Valid entries	Usage
y/n	If **y**, the Send-ANI backward signal corresponds exclusively to the caller-category request. In response to this signal, Avaya MultiVantage sends a forward signal containing the caller-category information on outgoing calls. On incoming calls, Avaya MultiVantage sends the Send-ANI backward signal upon receipt of the first address signal.

Restart ANI from Caller Category

Valid entries	Usage
y/n	If **y**, Avaya MultiVantage sends the caller-category signal later again when the signals for Caller-Category and ANI requests are the same and this signal is received after the Next-Digit forward signals have been received.

Number of Incoming ANI Digits

Valid entries	Usage
0 to **15**	Enter the number of ANI digits for incoming MFC calls.

Number of Outgoing ANI Digits

This field applies to Russian shuttle trunks, and MFC and MFE trunks.

Valid entries	Usage
0 to **15**	Enter the number of ANI digits for outgoing MFC calls. In India or any country where end-of-ani and end-of-digits are not defined for Tones to CO on Outgoing Forward Calls – Group I, Avaya MultiVantage appends ANI-Not-Available digits to ANI digits if the actual ANI length is less than the number entered in this field. If end-of-ani or end-of-digits are defined, this field is used in conjunction with Truncate Station Number in ANI as a maximum ANI length. For India, even if the length of ANI is defined, if the timeout occurs during the ANI collection, the call is routed with the ANI digits already collected.

Truncate Station Number in ANI

This field applies to Russian shuttle trunks, and MFC and MFE trunks.

Valid entries	Usage
beginning **ending** **no**	This field defines the side of the extension number from which to truncate when station ANI is sent to the CO and the combined length of the ANI prefix and extension number is greater than Number of Outgoing ANI Digits. The ANI prefix (either MFC or COR) is not truncated. There is no effect if ANI for PBX is sent.

ANI Source for Forwarded & Covered Calls

Valid entries	Usage
caller	Send the calling party’s ANI when calls are redirected.
forwarder	Send the forwarding party’s ANI when calls are redirected.

Address Digits Include End-of-Digits Signal

Indicates that an outgoing forward Group I end-of-digit signal is always sent after completion of address digits upon request from the Central Office for outgoing calls.

Valid entries	Usage
y/n	Enter **y** to send an outgoing forward Group I end-of-digit signal after completion of address digits upon request from the Central Office for outgoing calls.

Call Category for Vector ii-digits

Allows you to use the call category digit as the ii-digits on call vector steps.

Valid entries	Usage
y/n	If **y**, the call category digit, which is a part of ANI, is used as the ii-digits on call vector steps.

Request CPN at Start of Call

This field appears only if the Incoming Call Type field is **group-ii-mfc**. Provides for the switch to collect ANI and call category immediately after receipt of the first address digit.

Valid entries	Usage
y/n	If **y**, provides ANI (Calling Party Number (CPN) and call category) immediately after receiving the first address digit.

Do Not Send Group B Signals to CO

This field appears only if the Incoming Call Type field is **group-ii-mfc**. This field allows completion of a call without Group-B signals.

Valid entries	Usage
y	If **y**, does not send Group-B signals to complete an incoming call.
n	If **n**, sends Group-B signals to complete an incoming call.

ANI Available

Valid entries	Usage
1 to **15**	Enter the number of incoming and outgoing ANI-Available signals.

ANI Not Available

You must enter a value if the ANI for PBX field is blank

Valid entries	Usage
1 to **15**	Enter the number of the incoming and outgoing ANI-Not-Available signals. Your switch outpulses the End-of-Dial backward signal when the ANI-Not-Available forward signal is received on incoming calls. Your switch outpulses the ANI-Not-Available forward signal to the CO on outgoing calls where ANI is not possible.

Field descriptions for page 3

The fields shown on Page 3 of the Multifrequency-Signaling-Related System Parameters screen define the meaning of MFC tones for calls originated at the CO. Refer to ''Definitions of Group I, II, A, and B signals'' on page 1010 for more information.

When the screen initially appears, either of two sets of default values is possible. One set is for the group II call type; the other set is for non-group II call type. In each set, the default value for each field is set to the most common value.

Screen 210 shows the defaults when the Incoming Call Type field is **group-ii-mfc**. A variation appears if Incoming Call Type is **non-group-ii-mfc**. When Incoming Call Type is **non-group-ii-mfc**, group II and group B columns do not appear.

```
change system-parameters multifrequency-signaling                 Page 3 of 4
               MULTIFREQUENCY-SIGNALING-RELATED SYSTEM PARAMETERS

  INCOMING FORWARD SIGNAL TYPES           INCOMING BACKWARD SIGNAL TYPES
  (Tones from CO)                         (Tones to CO)

        Group-I             Group-II            Group-A             Group-B
   11: ignored            1: normal           1: next-digit        1: free
   12: ignored            2: normal           3: end-of-dial       2: busy
   13: ignored            3. normal          __:                   4: congestion
   14: ignored            4: normal          __: ________________  7: intercept
   15: ignored            5: normal          __: ________________ __: __________
                          6: normal          __: ________________ __: __________
                          7: normal          __: ________________ __: __________
                          8: normal          __: ________________ __: __________
                          9: normal          __: ________________ __: __________
                         10: normal          __: ________________ __: __________
                         11: normal          __: ________________ __: __________
                         12: normal          __: ________________ __: __________
                         13: normal          __: ________________ __: __________
                         14: normal          __: ________________ __: __________
                         15: normal          __: ________________ __: __________
```

Screen 210. Multifrequency-Signaling-Related System Parameters screen

Tones from CO on Incoming Forward Calls — Group I

Message codes 11 to 15 display. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) Assign a meaning to each code. Refer to ‘‘Definitions of Group I, II, A, and B signals’’ on page 1010 for signal type.

Valid entries	Usage
drop	If Incoming Call Type is **group-ii-mfc**
ani-avail	
end-of-ani	
end-of-dial	
ignored	
maint-call	
ani-not-avail	
send-congest	
drop	If the Incoming Call Type is **non-group-ii-mfc**
ignored	

Tones from CO on Incoming Forward Calls — Group II

Message codes 1 to 15 display. Assign a meaning to each code.

Valid entries	Usage
attendant	Refer to ''Definitions of Group I, II, A, and B signals'' on page 1010 for signal type.
busy-rt-attd	
data-call	
data-verify	
drop	
maint-call,	
send-intercept	
toll-auto	
toll-operator	
normal	

Tones from CO on Incoming Forward Calls — Group I

Message codes 11 to 15 display. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) Assign a meaning to each code.

Valid entries	Usage

Refer to ''Definitions of Group I, II, A, and B signals'' on page 1010 for signal type.

Tones to CO on Incoming Backward Calls — Group B

This field does not appear if the Do Not Send Group B Signals to CO field is **y**. Message codes between 1 and 15 display. Assign a meaning to each code.

Valid entries	Usage
busy	Refer to ''Definitions of Group I, II, A, and B signals'' on page 1010 for signal type.
congestion	
free	
mct	
tariff-free	
tie-free	
toll-busy	
intercept	

Field descriptions for page 4

The fields shown on this page define the meaning of MFC tones for calls originated at the PBX. Refer to ''Definitions of Group I, II, A, and B signals'' on page 1010 for more information.

Page 4 of the Multifrequency-Signaling-Related System Parameters screen only appears if Outgoing Call Type is **group-ii-mfc** or **mfe**.

```
change system-parameters multifrequency-signaling                  Page 4 of 4
                MULTIFREQUENCY-SIGNALING-RELATED SYSTEM PARAMETERS

  OUTGOING FORWARD SIGNAL TYPES           OUTGOING BACKWARD SIGNAL TYPES
  (Tones to CO)                           (Tones from CO)

        Group-I             Group-II            Group-A             Group-B
   11: ______________   2: normal         1: next-digit       1: free
   12: ______________   5: attendant      2: congestion       2: busy
   13: ______________   6: data-call      3: end-of-dial      3: congestion
   14: ______________  __: ______________ 4: congestion       4: congestion
   15: ______________  __: ______________ 5: call-info-ani    5: congestion
                       __: ______________ 6: congestion       6: free
                       __: ______________ 7: last-2-digits    7: intercept
                       __: ______________ 8: last-3-digits    8: congestion
                       __: ______________ 9: congestion       9: congestion
                       __: ______________10: congestion      10: congestion
                       __: ______________11: congestion      11: congestion
                       __: ______________12: congestion      12: congestion
                       __: ______________13: congestion      13: congestion
                       __: ______________14: congestion      14: congestion
                       __: ______________15: congestion      15: congestion
```

Screen 211. Multifrequency-Signaling-Related System Parameters screen

Tones to CO on Outgoing Forward Calls — Group I

Message codes 11 to 15 appear. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.)

Valid entries	Usage
end-of-digits	Assign a meaning to each code. Refer to ''Definitions of Group I, II, A, and B signals'' on page 1010 for signal type.
ani-avail,	
end-of-ani	
ani-not-avail.	

Tones to CO on Outgoing Forward Calls — Group II

Message codes between 1 and 15 display. Assign a meaning to each code. Each entry can only appear once in the group II column.

Valid entries	Usage
attendant	Refer to ''Definitions of Group I, II, A, and B signals'' on page 1010 for signal type.
data-call	
toll-auto	
normal	

Tones from CO on Outgoing Backward Calls — Group A

Message codes between 1 and 15 display. Assign a meaning to each code.

Valid entries	Usage
send-ani	Refer to ''Definitions of Group I, II, A, and B signals'' on page 1010 for signal type.
congestion	
drop	
end-of-dial	
last-2-digits	
last-3-digits	
last-digit	
next-ani-digit,	
next-digit	
restart	
intercept	
resend-digit	
setup-sppath	

Tones from CO on Outgoing Backward Calls — Group B

Message codes between 1 and 15 display. Assign a meaning to each code. Refer to ''Definitions of Group I, II, A, and B signals'' on page 1010 for signal type.

Valid entries	Usage
busy	
congestion	
free	
mct	
tariff-free	
toll-busy	
intercept	

Definitions of Group I, II, A, and B signals

Group I signals

Group I signals are a set of forward signals generated by the originating switch.

ani-avail

Used in Hungary. If this signal is defined and ANI is requested on outgoing R2-MFC calls, ANI is sent to the CO before ANI caller digits are sent. This signal is sent after the ANI caller category signal.

ani-not-avail

Used on DOD calls in Brazil and Columbia. The switch sends this signal to the CO when it receives an ANI request and the caller's number is not available.

digits 1 to 10

The signals from group I.1 to I.10 are reserved for address digits 0 to 9.

drop

When this signal is received from the CO, the switch starts the disconnect sequence and drops the call.

end-of-ani

This signal is used on DOD and DID calls. The switch sends this signal to indicate the end-of-ANI digits when ANI digits are sent to the CO.

end-of-dial

This signal is used when open numbering is used on DID calls. The CO sends this signal to indicate the end-of-dial digits and the switch responds with a request for a group II signal.

end-of-digits

This signal is sent by the origination switch that makes outgoing calls, sends digits, and receives a next-digit group A signal from the destination switch when there are no more digits to be sent.

This signal is also sent when the switch does not have end-of-ani assigned, makes an outgoing call, sends ANI, and receives a call-info-ani group A signal from the destination switch when there are no more ANI digits to be sent.

If both end-of-digits and end-of-ani are assigned, the switch uses end-of-ani after it sends the last ANI digit and end-of-digits after sending the last called-number digit.

ignored

If this signal is received from the CO, the switch sends a corresponding signal (A.1, and so on) but no action is taken in the response and it is not counted as a digit. In Belgium, this signal is not acknowledged.

maint-call

The CO sends a signal to indicate that a call is a maintenance call and the switch prepares the special maintenance call sequences for the CO. This signal may be used on DID calls in Saudi Arabia.

send-congestion

When the switch receives this signal from the CO on a DID call, it returns a congestion signal (group A), in compel (not pulse) mode, to the CO.

Group II signals

Group II signals are a more elaborate set of forward signals generated by the originating switch.

attendant

If the switch receives this signal on DID calls, the call terminates at an attendant regardless of the extension dialed. On DOD calls, this signal is sent to the CO if the CO requests calling-category information and the originating extension is an attendant. This signal is used on both DID and DOD calls.

busy-rt-attd

If the switch receives this signal on DID calls, the call terminates at an attendant if the called extension is busy or at the called extension if it is not busy. This signal is used on DID calls.

data-call

This signal is treated the same as the data-verify signal except that it does not require a terminating extension to be a data extension.

data-verify

If the switch receives this signal on DID calls and the terminating extension is not a data extension, it sends intercept treatment. On DOD calls, this signal is sent to the CO if the CO requests calling-category information and the originating extension is a data extension. This signal is used on both DID and DOD calls.

drop

When this signal is received from the CO, the switch starts the disconnect sequence and drops the call.

maint-call

The CO sends a signal to indicate that a call is a maintenance call and the switch prepares the special maintenance call sequences for the CO.

normal

This signal indicates that the caller is a normal subscriber. If it is received on a DID call, the call is terminated at the called extension. For an outgoing MF signaling call that uses group II signaling, this signal is sent to the CO when the CO requests calling-category information and the originating extension is a station. This signal is used in both DID and DOD calls.

send-intercept

If the switch receives this signal from the CO on a DID call, it returns group B intercept signal to the CO.

toll-auto

This signal is used in China. This signal indicates that a call is an automatic toll call. When the call terminates at a busy station and a special busy signal is defined, the busy signal is sent to the CO. You can define the special busy signal by choosing the option toll-busy on the incoming group B signals.

toll-operator

This signal, used in China, is treated as a normal subscriber signal. See the normal definition.

Group A signals

Group A signals are backward signals generated by the destination switch.

send-ani

The CO sends this signal to request calling-party category and sends additional signals to request ANI digits. This signal is sent to the CO when Avaya MultiVantage requests ANI digits on DID calls. This signal is used on both DOD and DID calls.

congestion

The CO sends this signal to indicate that it is experiencing network congestion. When the switch receives this signal on DOD calls, the switch drops the trunk and plays reorder tone to the calling party. This signal is used on DOD calls.

drop

When this signal is sent, the receiving switch starts the disconnect sequence.

end-of-dial

This signal is sent to indicate the end of the address digit string. For MF group II calls, this signal requests a group II signal and switches the sender over to the group B signaling mode. This signal is used on both DID and DOD calls.

intercept

The CO sends this signal to indicate the call has been terminated to an invalid destination. When the switch receives this signal on DOD calls, the switch drops the trunk and plays intercept tone to the calling party. This signal is used on DOD calls.

resend-digit

The switch sends this signal to adjust the outpulsing pointer so that the last digit can be resent again. This signal is used on DOD calls.

last-digit

The switch sends this signal to adjust the outpulsing pointer so that the last 2 digits can be resent. This signal is used on DOD calls.

last-2-digits

The switch sends this signal to adjust the outpulsing pointer so that the last 3 digits can be resent. This signal is used on DOD calls.

last-3-digits

The switch sends this signal to adjust the outpulsing pointer so that the last 4 digits can be resent. This signal is used on DOD calls.

next-digit

The switch sends this signal to request the next digit. This signal is used on both DID and DOD calls.

next-ani-digit

The switch sends this signal to request the next ANI digit. This signal is used on DID and DOD calls.

restart

The switch sends this signal to request the whole digit string again. This signal is used on DOD calls.

setup-sppath

The CO sends this signal to the switch to set up a speech path. This signal is used on DOD calls and on DID calls in Belgium.

Group B signals

Group B signals enhance group A signals for backward signaling from the destination switch by providing the status of the called party. In addition, if the originating switch uses group II signals, the destination switch answers with group B signals. Group B signals are as follows:

busy

This signal is sent to indicate that the called party is busy. On DID calls, the signal is sent to the CO if there is no coverage point to terminate the call. If the switch receives this signal on DOD calls, it plays busy tone to the calling party and drops the trunk.

congestion

This signal is sent to indicate that the system is congested and the call cannot be terminated successfully. On DID calls, the signal is sent to the CO to indicate that a resource is not available. On DOD calls, if the switch receives this signal, reorder tone is played to the calling party and the trunk is dropped.

free

This signal indicates that the called party is idle. On DID calls, the signal is sent to the CO to indicate that the called party is idle and the call is terminated successfully. If the switch receives this signal on DOD calls, it connects the trunk to the calling party.

intercept

This signal indicates that the called party number is not in service or is not correct. On DID calls, if intercept treatment is set to provide a tone, tone is sent to the CO to indicate that the called number is not valid. If the switch receives the signal on DOD calls, the switch plays intercept tone to the calling party and drops the trunk.

mct

This signal identifies the call as one that needs to be traced by the CO. Avaya MultiVantage then generates an MFC Call Trace Backward Signal (administered on the Multifrequency-Signaling-Related System-Parameters screen) during call setup instead of the “free” signal. If the terminating station’s COR has this feature set to **y,** the CO collects trace information before releasing the calling party.

> **NOTE:**
> If the station’s COR has MF Incoming Call Trace set to **y** and the “mct” signal is not defined, then the “free” signal is sent.

tariff-free

This signal is sent when the trunk group provides an “800” service. Avaya MultiVantage generates an MFC tariff-free backward signal (administered on the System-Parameters Multifrequency-Signaling screen) during call setup instead of the “free” signal, facilitating CO billing.

> **NOTE:**
> If the trunk is administered as a tariff-free trunk and the “tariff-free” signal is not defined, then the “free” signal is sent.

tie-free

This signal is used only when an incoming call is received and defined and the incoming facility is a tie trunk. Otherwise, the free signal is used.

toll-busy

This signal, used in China, is sent to indicate that the called party is busy if the call is an automatic toll call.

Music Sources

This screen defines music sources for Tenant Partitions. Each music source defined on the screen can be used by one or more Tenant Partitions. However, a partition may have only one music source.

NOTE:
If you use equipment that rebroadcasts music or other copyrighted materials, you may be required to obtain a copyright license from, or pay fees to, a third party such as the American Society of Composers, Artists, and Producers (ASCAP) or Broadcast Music Incorporated (BMI). You can purchase a Magic Hold® system, which does not require such a license, from Avaya.

Field descriptions for page 1

```
change music-source            Music Sources                      Page 1 of X

              Source    Type      Port       Description

                1      music      01A1003    Easy listening
                2      tone                  Tone-on-Hold
                3      music      01A1004    Rock
                4      none
                5      none
                6      none
                7      music      12B1301    Oldies
                8      none
                9      none
               10      none
               11      music      04C2003    Classical
               12      none
               13      none
               14      none
               15      none
```

Screen 212. Music Sources

Source

Display only field - the number assigned to this source. The maximum number of music sources is 20 for G3 csi and si and 100 for DEFINITY Server R. This screen appears with the appropriate pages to accommodate the number of music sources your system can support.

Type

If you entered a value in Music/Tone on Hold on the Feature-Related System Parameters screen, that value will appear in this field.

Valid entries	Usage
music	Enter the type of treatment to be provided by the music source.
tone	Only one music source may use this value.
none	

Port

Enter the auxiliary trunk or analog port address of the music source. Duplicates are not allowed. This field appears only if you entered **music** in Type.

Valid entries	Usage
1-x	cabinet
A-E	carrier
0-20	slot
01-31	circuit

Description

Enter a description of the administered music source. This field appears only if you entered **music** or **tone** in Type.

NOTE:
When Tenant Partitioning is enabled, Music/Tone on Hold on the Feature-Related System Parameters screen disappears. However, the value in that field (tone, music, or none) will appear as the first entry on the Music Sources screen. If the value was **music**, the port number also appears on the Music Sources screen. When Tenant partitioning is disabled, Music/Tone on Hold reappears on the Feature-Related System Parameters screen, along with and the values from the Music Sources screen.

Valid entries	Usage
20 alpha-numeric character (maximum)	

Packet Gateway Board

Use this screen to administer the Packet Gateway (PGATE) circuit pack.

NOTE:
The PGATE screen is used with DEFINITY Server R configurations.

Field descriptions for page 1

```
change pgate xxxxx
                           PACKET GATEWAY BOARD                  Page 1 of 1
        Board Location: _____                          Name: _______________
           Application:  X.25
   External cable type: rs232
    Port configuration: 1) rs232  2)rs232  3)rs232 4)rs232
```

Screen 213. Packet Gateway Board screen

Board Location

Enter the slot location of the PGATE circuit pack.

Valid entries	Usage
1-x	cabinet (Maximum value varies according to switch type)
A-E	carrier
0-20	slot

Name

Valid entries	Usage
Up to 15 alphanumeric characters	Enter the name of the adjunct with which the PGATE circuit pack communicates.

Application

Display-only field that shows the communications protocol used to transmit messages over the PGATE.

External Cable Type

Display-only field that shows the type of physical interface between the PGATE port and the adjunct.

Port Configuration

Display-only field that shows that the port is configured for “rs232” communication.

Partition Route Table

Use this table to identify routing for partition groups associated with an ARS analysis entry.

```
change partition route-table                                    Page 1 of X
                           Partition Routing Table

                             Routing Patterns
 Route
 Index         PGN 1    PGN 2    PGN 3    PGN 4    PGN 5    PGN 6    PGN 7    PGN 8
 -----         -----    -----    -----    -----    -----    -----    -----    -----
 196           ____     ____     ____     ____     ____     ____     ____     ____
 197           ____     ____     ____     ____     ____     ____     ____     ____
 198           ____     ____     ____     ____     ____     ____     ____     ____
 199           ____     ____     ____     ____     ____     ____     ____     ____
 200           ____     ____     ____     ____     ____     ____     ____     ____
 201           ____     ____     ____     ____     ____     ____     ____     ____
 202           ____     ____     ____     ____     ____     ____     ____     ____
 203           ____     ____     ____     ____     ____     ____     ____     ____
 204           ____     ____     ____     ____     ____     ____     ____     ____
 205           ____     ____     ____     ____     ____     ____     ____     ____
 206           ____     ____     ____     ____     ____     ____     ____     ____
 207           ____     ____     ____     ____     ____     ____     ____     ____
 208           ____     ____     ____     ____     ____     ____     ____     ____
 209           ____     ____     ____     ____     ____     ____     ____     ____
 210           ____     ____     ____     ____     ____     ____     ____     ____
```

Screen 214. Partition Route Table

PGN 1 (through PGN 8)

Enter the routing for each partition group associated with each route index number.

Valid entries	Usage
1 to **640**	Specifies the route pattern used to route the call
r1 to **r32**	Specifies the remote home numbering plan area table used to route the call
node	Designates node number routing
deny	Blocks the call

Personal CO Line Group

Use this screen to set up a personal central office line trunk group.

```
add personal-CO-line                                              Page 1 of x
                              PERSONAL CO LINE GROUP

 Group Number: __                    Group Type:_____________  CDR Reports: _
   Group Name: _________________                                      TAC: ____
Security Code: ____               Coverage Path: ____     Data Restriction? _
                               Outgoing Display? _

TRUNK PARAMETERS
                 Trunk Type: _____________            Trunk Direction: ________
                 Trunk Port: _______         Disconnect Timing(msec): ____
                 Trunk Name: __________             Trunk Termination: ______
         Outgoing Dial Type: _________              Analog Loss Group: ___
                   Prefix-1? _                     Digital Loss Group: ___
Disconnect Supervision - In? _                        Call Still Held? _
 Answer Supervision Timeout: ___          Receive Answer Supervision? _
                 Trunk Gain: ____                             Country: __
          Charge Conversion: _____             DS1 Echo Cancellation: _
              Decimal Point: ______
            Currency Symbol: ___
                Charge Type: _______
```

Screen 215.   Personal CO Line Group

The following fields are unique to this screen.

Analog Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over an analog signaling port in the trunk group.

Valid entries	Usage
1 to **17**	Shows the values from the loss plan and tone plan.

Coverage Path

Valid entries	Usage
1 to **999**	Enter the number of the call coverage path you want to use for incoming calls.
t1 to **t999**	Enter the number of a time-of-day table.
blank	Assigning a coverage path is optional: leave this field blank if you do not want to assign one.

Digital Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over a digital signaling port in the trunk group.

Valid entries	Usage
1 to **17**	Shows the values from the loss plan and tone plan.

DS1 Echo Cancellation

Allows you to administer echo cancellation per channel.

Valid entries	Usage
y/n	Enter **y** to activate echo cancellation capability.

Security Code

Valid entries	Usage
4 digits	Enter a 4-digit code that users must dial to retrieve voice messages and to use the Demand Print Message feature.
blank	Leave this field blank if you do not want to use a security code to control access.

Trunk Direction

Valid entries	Usage
incoming **outgoing** **two-way**	Enter the direction of the traffic on this trunk group. The entry in this field affects which timers appear on the Administrable Timers page. For WATS Group Types, only **incoming** or **outgoing** may be entered.

Trunk Port

Valid entries	Usage
port address	Enter the full address of the port for this trunk group.

Trunk Name

Valid entries	Usage
1 to 10 characters blank	Enter a descriptive name for this trunk. Don’t use the group type (CO, FX, WATS) here. For example, you might use names that identify the vendor and function of the trunk group: USWest Local; Sprint Toll, etc.

Field descriptions for page 2

```
add personal-CO-line                                              Page 1 of x
                           PERSONAL CO LINE GROUP

ASSIGNED MEMBERS (Stations with a button for this PCOL Group)

        Ext     Name                            Ext     Name

     1:                                     9:
     2:                                    10:
     3:                                    11:
     4:                                    12:
     5:                                    13:
     6:                                    14:
     7:                                    15:
     8:                                    16:
```

Screen 216. Personal CO Line Group

Ext

This display-only field shows the extension of phones that have a CO Line button.

Name

This display-only field shows the name assigned to phones that have a CO Line button.

Field descriptions for page 3

Administrable timers for Personal CO Line groups appear on Field descriptions for page 3. Refer to ‘‘Trunk Group’’ on page 1167 for standard field definitions of the available timers.

Related topics

Refer to ‘‘Adding a PCOL trunk group’’ on page 412 for instructions.

Refer to ‘‘Trunk Group’’ on page 1167 for definitions of all trunk group fields that are *not* unique to the PCOL screen.

Pickup Group

This screen implements call pickup groups with up to 50 extensions per group. A pickup group is a group of users authorized to answer calls to a phone extension within that group of users. A phone extension number can only belong to one pickup group.

Field descriptions for pages 1 and 2

```
change pickup-group 1                                        Page   1 of   2
                                  PICKUP GROUP

                   Group Number: 1        Extended Group Number: ____

GROUP MEMBER ASSIGNMENTS

   Ext      Name (first 26 characters)      Ext      Name (first 26 characters)
 1:51001    mips 2a1501                     14:
 2:                                         15:
 3:                                         16:
 4:                                         17:
 5:                                         18:
 6:                                         19:
 7:                                         20:
 8:                                         21:
 9:                                         22:
10:                                         23:
11:                                         24:
12:                                         25:
13:
```

Screen 217. Pickup Group

Group Number

A display-only field when the screen is accessed using an administration command such as **add** or **change**.

Valid entries	Usage
	Enter a Pickup Group number when completing a paper screen.

Extended Group Number

This field appears only when the Group Call Pickup field is set to **flexible** on the Feature-Related System Parameters screen. The extended group is a collection of pickup groups that can answer calls from other pickup groups in the same extended group.

Valid entries	Usage
1-100 (DEFINITY Server SI and DEFINITY Server CSI)	Enter the extended group number or blank.
1-400 (DEFINITY Server R)	

Ext

Enter the extension assigned to a station.

Valid entries	Usage
An extension number.	A VDN cannot be assigned to a Call Pickup group.

Name

This display-only field shows the name assigned to the above extension number when the users and their associated extensions were administered.

PRI Endpoint

This screen administers PRI Endpoints for the Wideband Switching feature.

NOTE:
A PRI Endpoint with a width greater than 1 may be administered only if the Wideband Switching feature has been enabled on the System-Parameters Customer-Options screen.

A PRI Endpoint is an endpoint application connected to line-side ISDN-PRI facilities and has standard ISDN-PRI signaling interfaces to the system. For information on endpoint applications connected to line-side non-ISDN T1 or E1 facilities, see ‘‘Access Endpoint’’ on page 575 in this module.

A PRI Endpoint is defined as 1–31 adjacent DS0s/B-channels, addressable via a single extension, and signaled via a D-channel (Signaling Group) over a standard T1 or E1 ISDN-PRI interface.

```
add pri-endpoint next                                          Page   1 of   1
                                  PRI ENDPOINT

                     Extension: 300
                          Name: 27 character PRI Endpoint 1
               (Starting) Port:                          Width: 1
  Originating Auto Restoration? n             Signaling Group:
                           COR: 1                          COS: 1
                            TN: 1          Simultaneous Calls? n
            Maintenance Tests? y

                             WIDEBAND SUPPORT OPTIONS

                                H0? n
                               H11? n
                               H12? n
                             NXDS0? y   Contiguous? n
```

Screen 218. PRI Endpoint

Extension

A display-only field when the screen is accessed using an administration command such as **change** or **display**.

Valid entries	Usage
blank	This is the extension number used to access the PRI endpoint. Enter a valid unassigned extension number when completing a paper screen.

Name

Identifies the endpoint.

Valid entries	Usage
Up to 27 alphanumeric characters.	

(Starting) Port

Enter the seven-character starting port of the PRI Endpoint.

Valid entries	Usage
01 through **22** (DEFINITY Server R)	First and second characters are the cabinet number
01 through **03** (DEFINITY Server SI)	
01 through **02** (G3s)	
A through **E**	Third character is the carrier
01 through **20**	Fourth and fifth characters are the slot number
01 through **31** (DS1 Interface ports)	Sixth and seventh characters are the circuit number

Width

Enter the number of adjacent DS0 ports beginning with the specified Starting Port, that make up the PRI Endpoint.This field cannot be blank.

Valid entries	Usage
1 to **31**	A width of **6** defines a PRI Endpoint that can support data rates up to 384 Kbps.

Originating Auto Restoration

Valid entries	Usage
y/n	Enter **y** to automatically restore calls originating from this PRI Endpoint (while maintaining endpoint call status) in the case of network failure if the call is over SDDN network facilities.

Signaling Group

Valid entries	Usage
1 to **30**	Enter the D-channel or D-channel pair that will provide the signaling information for the set of B-channels that make up the PRI Endpoint.

COR

Valid entries	Usage
0 to **95**	Enter class of restriction (COR) to determine calling and called party privileges

COS

Valid entries	Usage
0 to **15**	Enter the Class of Service (COS) to determine the features that can be activated by, or on behalf of, the endpoint.

TN

Valid entries	Usage
1–**20**	Enter the Tenant Partition number.

Simultaneous Calls

Valid entries	Usage
y/n	Enter **y** to specify that multiple simultaneous calls can be placed to/from the PRI Endpoint.

Maintenance Tests

Valid entries	Usage
y/n	Enter **y** to run hourly maintenance tests on this PRI Endpoint.

H0

Valid entries	Usage
y/n	Enter **y** to specify the ISDN information transfer rate for 384 Kbps of data, which is comprised of six B-channels. When a PRI Endpoint is administered to support H0, the hunt algorithm to satisfy a call requiring 384 Kbps of bandwidth uses a fixed allocation scheme.

H11

Valid entries	Usage
y/n	Enter **y** to specify the ISDN information transfer rate for 1536 Kbps of data, which is comprised of 24 B-channels. When a PRI Endpoint is administered to support H11, the hunt algorithm to satisfy a call requiring 1536 Kbps of bandwidth uses a fixed allocation scheme.

H12

Valid entries	Usage
y/n	Enter **y** to specify the ISDN information transfer rate for 1920 Kbps data, which includes 30 B-channels. When a PE is administered to support H12, the hunt algorithm to satisfy a call requiring 1920 Kbps of bandwidth uses a fixed allocation scheme.

NXDS0

Valid entries	Usage
y/n	Enter **y** to specify the NXDS0 multi-rate service.

Contiguous

Specifies whether to hunt contiguous NXDS0 channels. This field only appears if **y** is entered in NXDS0. The hunt algorithm to satisfy an NXDS0 call is as follows:

Valid entries	Usage
y	Enter **y** to specify the "floating" scheme. NXDS0 calls are placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without constraint on the starting channel (no fixed starting point trunk). H0 and NXDS0 "floating" scheme cannot both be **y**.
n	Enter **n** to specify the "flexible" scheme. NXDS0 calls are placed on any set of B-channels on the same facility as long as the requested bandwidth is satisfied. There are no constraints, such as contiguity of B-channels or fixed starting points.

QSIG to DCS TSC Gateway screen

The QSIG to DCS TSC Gateway screen determines when and how to convert messages from a QSIG NCA-TSC to an administered AUDIX NCA-TSC. This screen maps the QSIG subscriber number to the appropriate AUDIX signaling group and TSC index.

This screen only appears if the Interworking with DCS field is **y** on the System-Parameters Customer-Options screen.

```
change isdn qsig-dcs-tsc-gateway                                    Page 1 of 1

                            QSIG TO DCS TSC GATEWAY

      Subscriber        Sig   TSC          Subscriber         Sig       TSC
       Number           GRP  Index          Number            GRP      Index
    ____________________ ___   ___       ____________________  ___        ___
    ____________________ ___   ___       ____________________  ___        ___
    ____________________ ___   ___       ____________________  ___        ___
    ____________________ ___   ___       ____________________  ___        ___
    ____________________ ___   ___       ____________________  ___        ___
    ____________________ ___   ___       ____________________  ___        ___
    ____________________ ___   ___       ____________________  ___        ___
    ____________________ ___   ___       ____________________  ___        ___
    ____________________ ___   ___       ____________________  ___        ___
    ____________________ ___   ___       ____________________  ___        ___
    ____________________ ___   ___       ____________________  ___        ___
    ____________________ ___   ___       ____________________  ___        ___
    ____________________ ___   ___       ____________________  ___        ___
```

Screen 219. QSIG to DCS TSC Gateway screen

Subscriber Number

You can enter up to 28 subscriber numbers.

Valid entries	Usage
0 - 9, *****, ‘**x**’, ‘**X**’	Enter a subscriber number up to 20 characters in length. You can use wildcards ‘x’ and ‘X’ to enter subscriber numbers.

Sig Grp

Valid entries	Usage
1 - 110	Enter the assigned signaling group number between 1 and 110 for DEFINITY Server SI
1 - 416	Enter the assigned signaling group number between 1 and 416 for DEFINITY Server R

TSC Index

You must complete the TSC Index field for each machine ID.

Valid entries	Usage
1 - 64	Enter the assigned signaling group number for **qsig-mwi** application type on the Signaling Group screen.

Voice Mail Number

This field can be left blank.

Valid entries	Usage
0 - 9	Enter the complete Voice Mail Dial Up number up to 15 digits.

AAR/ARS Access Code

This field can be left blank.

Valid entries	Usage
0 - 9, *, #	Enter up to 4-digit access code.

Remote Access

Valid entries	Usage
1 - 64	Enter the assigned signaling group number for **qsig-mwi** application type on the Signaling Group screen.

The Remote Access screen is used to implement the Remote Access feature. Remote Access permits a caller located outside the system to access the system through the public or private network and then use the features and services of the system.

Remote Access users can dial into the system using central office (CO), Foreign Exchange (FX), Wide Area Telecommunications trunks (WATS), and Integrated Services Digital Network Primary Rate Interface (ISDN-PRI) trunks. In addition, a dedicated Remote Access Direct Inward Dialing number can be provided.

SECURITY ALERT:

Avaya designed the Remote Access feature incorporated in this product that, when properly administered by the customer, enables the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of remote access features. In such an event, applicable tariffs require the customer pay all network charges for traffic. Avaya cannot be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

To ensure the security of your system, consider the following:

- Make all remote access facilities unlisted directory telephone numbers.
- Require users to enter a Barrier Code of at least seven random digits AND an Authorization Code of at least 13 random digits to make network calls.
- Make Authorization Codes nonconsecutive (random) and change them, at least, quarterly.
- Deactivate Authorization Codes immediately if the user leaves the company or changes assignments.

- Assign the minimum level of calling permissions required to each Authorization Code.
- Block off-hours and weekend remote access calling, when possible. Use Alternative Facility Restriction Levels, if available.
- Use a voice recording, warble tone, or no tone and avoid use of a dial tone as a prompt when the remote access unit answers.
- Assign the lowest possible FRL to only allow internal switch calls.

As an additional step to ensure System security, you can permanently disable the Remote Access feature if you do not intend to use it now or in the future. If you do decide to permanently disable the feature, it will require Avaya Services intervention to activate the feature again.

CAUTION:

Your attempt to disable the Remote Access feature will be lost if the switch is rebooted without saving translations. Therefore, execute a "save translation" command after permanently disabling the Remote Access feature.

To assist you in maintaining the security of your system, Avaya MultiVantage provides the status remote access command, which provides status and information on each remote access barrier code and on the remote access feature. A sample Status Remote Access screen follows the Remote Access screen.

Field descriptions for page 1

```
change remote-access

                                  REMOTE ACCESS
   Remote Access Extension: ________             Barrier Code Length:____
Authorization Code Required? y              Remote Access Dial Tone: n

     Barrier      COR  TN  COS   Expiration    No. of     Calls
      Code        COR  TN  COS      Date       Calls      Used
   1:________     1__  1_  1__   __/__/__      ______     ______
   2:________     1__  1_  1__   __/__/__      ______     ______
   3:________     1__  1_  1__   __/__/__      ______     ______
   4:________     1__  1_  1__   __/__/__      ______     ______
   5:________     1__  1_  1__   __/__/__      ______     ______
   6:________     1__  1_  1__   __/__/__      ______     ______
   7:________     1__  1_  1__   __/__/__      ______     ______
   8:________     1__  1_  1__   __/__/__      ______     ______
   9:________     1__  1_  1__   __/__/__      ______     ______
  10:_______     1__  1_  1__   __/__/__      ______     ______
Permanently Disable? __ Disable Following A Security Violation? y
(NOTE:  You must logoff to effect permanent disabling of Remote Access)
```

Screen 220. Remote Access

```
status remote-access

                              REMOTE ACCESS STATUS

                     Remote Access Status: enabled
                       Date/Time Modified: 01/30/95 17:00

   Barrier  Date     Expiration No. of Calls Status Date/Time Expired Cause
   Code     Modified Date       Calls  Used
 1:2374745  01/30/95 03/31/95   50     50    expired 02/15/95 20:43   calls
 2:3374837  01/30/95   /  /     20     4     active    /  /
 3:3285038  01/30/95 01/31/96          13    expired 02/10/95 09:32   date
 4:5738557  01/30/95 07/31/95   20     20    expired 02/03/95 10:14   calls
 5:7764884  01/30/95 05/20/95          0     active    /  /      :
 6:           /  /     /  /                            /  /      :
 7:           /  /     /  /                            /  /      :
 8:           /  /     /  /                            /  /      :
 9:           /  /     /  /                            /  /      :
10:           /  /     /  /                            /  /      :
```

Screen 221. Remote Access Status

Remote Access Extension

The remote access extension is used as if it was a DID extension. Only one DID extension can be assigned as the remote access extension. Calls to that number are treated the same as calls on the remote access trunk.

When a trunk group is dedicated to Remote Access, the remote access extension number is administered on the trunk group's incoming destination field.

Valid entries	Usage
Extension number	Enter the extension number for Remote Access associated with each trunk that supports the Remote Access feature. You cannot assign a Vector Directory Number (VDN) extension as the remote access extension.

Barrier Code Length

Assign a barrier code length of 7 to provide maximum security.

Valid entries	Usage
4 to **7**	Enter a number to indicate the length of the barrier code.

Authorization Code Required

When you use an authorization code in conjunction with a barrier codes it increases the security of the Remote Access feature.

Valid entries	Usage
y/n	Enter **y** to require an authorization code be dialed by Remote Access users to access the system's Remote Access facilities.

Remote Access Dial Tone

Set this field to **n** for maximum security. This field appears when the Authorization Code Required field is set to y.

Valid entries	Usage
y/n	Enter **n** so that there is no Remote Access Dial Tone prompt.

Barrier Code

You must assign a barrier code that conforms to the number entered in the Barrier Code Length field. You may enter up to 10 barrier codes per system. Duplicate entries are not allowed. You must keep your own records regarding the distribution of these barrier codes to your personnel.

Valid entries	Usage
0 to **9**	Enter a 4- to 7-digit number in any combination of digits.
none	Must be specified in the first Barrier Code field, if the Barrier Code Length field is blank.

COR

Assign the most restrictive class of restriction (COR), that provides only the level of service required, to provide the maximum security.

Valid entries	Usage
0 through **95**	Enter the COR number associated with the barrier code that defines the call restriction features.

TN

Valid entries	Usage
1 to **20** (DEFINITY Server SI)	Enter the appropriate Tenant Partition number.
1 to **100** (DEFINITY Server R)	

COS

Assign the most restrictive class of service (COS), that provides only the level of service required to provide the maximum security.

Valid entries	Usage
0 to **15**	Enter the COS number, associated with the barrier code, that defines access permissions for Call Processing features.

Expiration Date

Assign an expiration date based on the expected length of time the barrier code will be needed. If it is expected the barrier code is to be used for a 2-week period, assign a date two weeks from the current date. If the Expiration Date is assigned, a warning message is displayed on the system copyright screen seven days prior to the expiration date. The system administer can modify the expiration date to extend the time interval if needed.

Valid entries	Usage
A date greater than the current date	Enter the date you want the barrier code to expire.

No. of Calls

The Expiration Date and No. of Calls fields can be used independently or in conjunction to provide the maximum security. If both the Expiration Date and No. of Calls fields are assigned, the corresponding barrier code expires when the first of these criteria is satisfied.

Valid entries	Usage
1 to **9999**	Enter the number of Remote Access calls that can be placed using the associated barrier code.

Calls Used

This display-only field shows the number of calls placed using the corresponding barrier code. This field is incremented each time a barrier code is successfully used to access the Remote Access feature. A usage that exceeds the expected rate indicates improper use.

Permanently Disable

Reactivation of remote access to the interface requires Avaya Services intervention.

Valid entries	Usage
y/n	Enter **y** to permanently block remote access to the administration interface.

Disable Following a Security Violation

This field appears on the screen when the SVN Authorization Code Violation Notification Enabled field on the Security-Related System Parameters screen is set to y.

Valid entries	Usage
y/n	Enter **y** to disable the remote access feature following detection of a remote access security violation. The system administrator can re-enable Remote Access using the enable remote access command.

Related Topics

Refer to ''Setting up remote access'' on page 364 for step-by-step instructions for configuring remote access.

Refer to ''Remote Access'' on page 1680 for a description of this feature.

Remote Call Coverage Table

The Remote Call Coverage Table allows you to provide automatic redirection of certain calls to alternate non-local answering positions in a coverage path.

Non-local numbers may be any ARS or AAR number, any number on the public network, any international number, or a UDP/DCS extension up to 16 digits, which includes any ARS/AAR facility access code, any trunk dial access code (TAC), long distance dialing code, or international dial code. Up to 999 remote call coverage points can be entered on the 23 pages of this screen.

Field descriptions for page 1

```
change coverage remote

                       REMOTE CALL COVERAGE TABLE
01: ________________      16: ________________      31: ________________
02: ________________      17: ________________      32: ________________
03: ________________      18: ________________      33: ________________
04: ________________      19: ________________      34: ________________
05: ________________      20: ________________      35: ________________
06: ________________      21: ________________      36: ________________
07: ________________      22: ________________      37: ________________
08: ________________      23: ________________      38: ________________
09: ________________      24: ________________      39: ________________
10: ________________      25: ________________      40: ________________
11: ________________      26: ________________      41: ________________
12: ________________      27: ________________      42: ________________
13: ________________      28: ________________      43: ________________
14: ________________      29: ________________      44: ________________
15: _____________         30: _____________         45: _____________
```

Screen 222. Remote Call Coverage Table

RHNPA Table

The RHNPA Table defines route patterns for specific 3-digit codes, usually direct distance dialing (DDD) prefix numbers.

```
change rhnpa table                                                     Page 1 of X
                                 RHNPA TABLE: __
                        CODE:  x00 - x99
                              Pattern Choices
         1: ___   3: ___   5: ___   7: ___   9: ___   11: ___
         2: ___   4: ___   6: ___   8: ___  10: ___   12: ___
     Code-Pattern Choice Assignments (from 1 - 12 above)
00: 1__ 10: 1__ 20: 1__ 30: 1__ 40: 1__ 50: 1__ 60: 1__ 70: 1__ 80: 1__ 90: 1__
01: 1__ 11: 1__ 21: 1__ 31: 1__ 41: 1__ 51: 1__ 61: 1__ 71: 1__ 81: 1__ 91: 1__
02: 1__ 12: 1__ 22: 1__ 32: 1__ 42: 1__ 52: 1__ 62: 1__ 72: 1__ 82: 1__ 92: 1__
03: 1__ 13: 1__ 23: 1__ 33: 1__ 43: 1__ 53: 1__ 63: 1__ 73: 1__ 83: 1__ 93: 1__
04: 1__ 14: 1__ 24: 1__ 34: 1__ 44: 1__ 54: 1__ 64: 1__ 74: 1__ 84: 1__ 94: 1__
05: 1__ 15: 1__ 25: 1__ 35: 1__ 45: 1__ 55: 1__ 65: 1__ 75: 1__ 85: 1__ 95: 1__
06: 1__ 16: 1__ 26: 1__ 36: 1__ 46: 1__ 56: 1__ 66: 1__ 76: 1__ 86: 1__ 96: 1__
07: 1__ 17: 1__ 27: 1__ 37: 1__ 47: 1__ 57: 1__ 67: 1__ 77: 1__ 87: 1__ 97: 1__
08: 1__ 18: 1__ 28: 1__ 38: 1__ 48: 1__ 58: 1__ 68: 1__ 78: 1__ 88: 1__ 98: 1__
09: 1__ 19: 1__ 29: 1__ 39: 1__ 49: 1__ 59: 1__ 69: 1__ 79: 1__ 89: 1__ 99: 1__
```

Screen 223. RHNPA Table

RHNPA TABLE

Valid entries	Usage
1 to **32**	Enter the table number. You can use up to 8 screens for each table, one screen for each 100 numbers.

CODE

Enter the desired 100-block, for example 000 through 099 or 900 through 999. A separate screen is required for each 100-block.

Pattern Choices

Valid entries	Usage
	Enter the route pattern number you want associated with each code. The pattern choice you list on one screen automatically defaults to the other screens of the same table. If you use one pattern for most of the codes, assign that pattern to choice 1.

Code-Pattern Choice Assignments

Valid entries	Usage
1 through **12**	Enter a Pattern Choice number from the list above to choose the route pattern for each prefix code.

Route Pattern

The Route Pattern screen defines the route patterns used by your switch. Each route pattern contains a list of trunk groups that can be used to route the call. The maximum number of route patterns and trunk groups allowed depends on the configuration and memory available in your system.

Use this screen to insert or delete digits so AAR or ARS calls route over different trunk groups. You can convert an AAR number into an international number, and insert an area code in an AAR number to convert an on-network number to a public network number. Also, when a call directly accesses a local central office (CO), if the long-distance carrier provided by your CO is not available, your switch can insert the dial access code for an alternative carrier into the digit string.

```
change route-pattern 1                                                Page 1 of X
                              Pattern Number: 1_

                              No.                                          DCS/
   Grp. FRL NPA Pfx Hop Toll Del   Inserted                               QSIG  IXC
   No.          Mrk Lmt List Dgts  Digits                                  Intw
1: ___   _ ___ _ __   __   __   _____________________________________   n   user
2: ___   _ ___ _ __   __   __   _____________________________________   n   user
3: ___   _ ___ _ __   __   __   _____________________________________   n   user
4: ___   _ ___ _ __   __   __   _____________________________________   n   user
5: ___   _ ___ _ __   __   __   _____________________________________   n   user
6: ___   _ ___ _ __   __   __   _____________________________________   n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature BAND  No.   Numbering LAR
    0 1 2 3 4 W     Request                                 Dgts   Format
                                                        Subaddress
 1: y y y y y n  y  none  ___ both ept outwats-bnd____ ____  _    ________  none
 2: y y y y y n  y            rest     _______________       _    ________  next
 3: y y y y y n  y            rest     _______________       _    ________  rehu
 4: y y y y y n  y            rest     _______________       _    ________  none
 5: y y y y y n  y            rest     _______________       _    ________  none
 6: y y y y y n  y            rest     _______________       _    ________  none
```

Screen 224. Route Pattern

Pattern Number

Displays the route pattern number (**1** to **640**).

Grp No

Valid entries	Usage
1 to **99**	Enter the trunk group number associated with this row (preference).

FRL

Valid entries	Usage
0 to **7**	Enter the Facility Restriction Level (FRL) associated with the entries on this row (preference). 0 is the least restrictive and 7 is the most restrictive. The calling party's FRL must be greater than or equal to this FRL to access the associated trunk-group.

SECURITY ALERT:

For system security reasons, Avaya recommends using the most restrictive FRL possible.

Network Specific Facility

Identifies the services and features used to complete a call.

NPA

This entry is not required for AAR.

Valid entries	Usage
3-digit number	Enter the 3-digit Numbering Plan Area (NPA) (or area code) for the terminating endpoint of the trunk group. Call your local telephone company to verify this number if you need help. For WATS trunks, the terminating NPA is the same as the home NPA unless the Local Exchange Carrier requires 10 digits for local NPA calls.
blank	For AAR calls and for tie trunks

Prefix Mark

This entry is not required for AAR. For ARS, enter a number from **0** to **4** or blank.

Prefix Marks set the requirements for sending a prefix digit 1, indicating a long-distance call. Prefix Marks apply to 7- or 10-digit Direct Distance Dialing (DDD) public network calls. A prefix digit 1 is sent only when call type is foreign number plan area (FNPA) or home numbering plan area (HNPA) in the ARS Digit Analysis table.

For a WATS trunk, the Prefix Mark is the same as the local CO trunk.

Valid entries	Usage
0	■ Suppress a user-dialed prefix digit 1 for 10-digit FNPA calls. ■ Leave a user-dialed prefix digit 1 for 7-digit HNPA calls. ■ Leave a prefix digit 1 on 10-digit calls that are not FNPA or HNPA calls. Do not use Prefix Mark 0 in those areas where all long-distance calls must be dialed as 1+10 digits. Check with your local network provider.
1	■ Send a 1 on 10-digit calls, but not on 7-digit calls. Use Prefix Mark 1 for HNPA calls that require a 1 to indicate long-distance calls.
2	■ Send a 1 on all 10-digit and 7-digit long-distance calls. Prefix Mark 2 refers to a Toll Table to define long distance codes.
3	■ Send a 1 on all long-distance calls and keep or insert the NPA (area code) so that all long distance calls are 10-digit calls. The NPA is inserted when a user dials a Prefix digit 1 plus 7-digits. Prefix Mark 3 refers to a Toll Table to define long distance codes.
4	■ Always suppress a user-dialed Prefix digit 1. Use Prefix Mark 4, for example, when ISDN calls route to a switch that rejects calls with a prefix digit 1.
blank	For tie trunks, leave this field blank.

Hop Lmt

Enter the number of hops for each preference. A hop is when a call tandems through a switch to another destination. Limiting the number of hops prevents circular hunting, which ties up trunk facilities without ever completing the call. Avaya MultiVantage blocks a hop equal to or greater than the number you enter.

Valid entries	Usage
blank	Indicates that there is no limit to the number of hops for this preference.
1 to **9**	To limit the number of hops if using the tandem hop feature.
1 to **32**	If using the transit feature.

Toll List

This entry is not required for AAR.

Valid entries	Usage
1 to **32**	For ARS, enter the number of the ARS Toll Table associated with the terminating NPA of the trunk group. You must complete this field if Prefix Mark is **2** or **3**.

No. Del. Digits

Use this field to modify the dialed number so an AAR or ARS call routes over different trunk groups that terminate in switches with different dial plans

Valid entries	Usage
0 to **28**	Enter the total number of digits you want the system to delete before it sends the number out on the trunk. Use for calls that route: ■ to or through a remote switch ■ over tie trunks to a private network switch ■ over Central Office (CO) trunks to the serving CO

Inserted Digits

Enter the digits you want inserted for routing. The switch can send up to 52 digits. This includes up to 36 digits you may enter here plus up to 18-digits originally dialed. Special symbols count as two digits each.

Valid entries	Usage
(**0** to **9**)	Enter the digits you want inserted for routing.
0 to 36 digits	
*	When * is in the route pattern and the outgoing trunk is signaling type “mf”, the MFC tone for the “end-of-digits” is sent out to the CO in place of the *.
#	When # is in the route pattern and the outgoing trunk is signaling type “mf”, the MFC tone for the “end-of-digits” is sent out to the CO in place of the #.
‘**,**’	Use 2 places. Creates a 1.5 second pause between digits being sent. Do not use as the first character in the string unless absolutely necessary. Misuse can result in some calls, such as Abbreviated Dialing or Last Number Dialed, not completing.

Valid entries	Usage
+	Wait for dial tone up to the Off Premises Tone Detection Timer and then send digits or intercept tone based on Out Pulse Without Tone y/n on the Feature-Related System Parameters screen.
%	Start End-to-End Signaling.
!	Wait for dial tone without timeout and then send DTMF digits.
&	Wait for ANI (used for Russian pulse trunks)

DCS/QSIG Intw

This field only appears if the Interworking with DCS field on the System-Parameters Customer-Options screen is set to **y**.

Valid entries	Usage
y/n	Enter **y** to enable DCS/QSIG Interworking.

IXC

Inter-Exchange Carrier (IXC) identifies the carrier, such as AT&T, used for calls that route via an IXC, and for Call Detail Recording (CDR).

This field appears when the ISDN-PRI or ISDN-BRI Trunks field is **y** on the System-Parameters Customer-Options screen.

Valid entries	Usage
Valid carrier code	Identifies the carrier for IXC calls
user	For presubscribed carrier. Used when an IXC is not specified.
none	This field must be **none** for non-ISDN trunk groups and for Bellcore NI-2 Operator Service Access. If you need to send an IXC code for a non-ISDN trunk group, enter the IXC code in the Inserted Digits field.

BCC Value

Bearer Capability Class (BCC) identifies the type of call appropriate for this trunk group, such as voice calls and different types of data calls. This field appears when the ISDN-PRI or ISDN-BRI Trunks field is **y** on the System-Parameters Customer-Options screen.

Valid entries	Usage
y/ **n**	Enter **y** in appropriate BCC column (0, 1, 2, 3, 4, or W) if the BCC is valid for the associated route pattern. A trunk group preference may have more than one BCC.

The following table explains BCC values:

BCC Value	Description
0	Voice-Grade Data and Voice
1	56-kbps Data (Mode 1)
2	64-kbps Data (Mode 2)
3	64-kbps Data (Mode 3)
4	64-kbps Data (Mode 0)
W	128 to 1984-kbps Data (Wideband)

TSC

Set TSC to **y** for feature transparency on DCS+ calls and to use QSIG Call Completion.

Valid entries	Usage
y/n	Enter **y** to allow Call-Associated TSCs, and to allow incoming Non-Call-Associated TSC requests to be tandemed out for each preference.

CA-TSC Request

Use CA-TSC on ISDN B-channel connections.

Valid entries	Usage
as-needed	The CA-TSC is set up only when needed. This causes a slight delay. This entry is recommended for most situations.
at-setup	The CA-TSC is automatically set up for every B-channel call whether or not it is needed.
none	No CA-TSC is set up. Permits tandeming of NCA-TSC setup requests0.

ITC (Information Transfer Capability)

Use Information Transfer Capability (ITC) to identify the type of data transmission or traffic that this routing preference can carry. The ITC applies only to data calls (BCC 1 through 4).

This field must be **unre** or **both** if the BCC is **y** and the BCC value is **W**

Valid entries	Usage
both	Calls from restricted and unrestricted endpoints can access the route pattern.
rest(ricted)	Calls from restricted endpoints can access the route pattern.
unre(stricted)	Calls from unrestricted endpoints can access the route pattern.

BCIE (Bearer Capability Information Element)

Use BCIE to determine how to create the ITC codepoint in the setup message. This field applies to ISDN trunks and appears if ITC is **both**.

Valid entries	Usage
ept (endpoint)	
unr (unrestricted)	

Service/Feature

Enter up to 15 characters to identify the Service/Feature carried by the information element (IE) in a call in this route pattern. This field is required by Call-by-Call Service Selection, and Network Call Redirection and Transfer.

Service/Feature appears when ISDN-PRI or ISDN-BRI Trunks is **y** on the System-Parameters Customer-Options screen.

Valid entries		
accunet	**multiquest**	**sdn** (Enter to allow Network Call Redirection/Transfer)
i800	**operator**	**sub-operator**
inwats	**oper-lds** (operator and lds)	**sub-op-lds** (sub-operator and lds)
lds	**oper-meg** (operator and megacon)	**sub-op-meg** (sub-operator and megacom)
mega800	**oper-sdn** (operator and sdn)	**sub-op-sdn** (sub-operator and sdn)
megacom	**outwats-bnd**	**wats-max-bnd**

Band

Enter a number that represents the OUTWATS band number (US only).

WATS is a voice-grade service that provides both voice and low-speed data transmission calls to defined areas (bands) for a flat rate charge.0

This field appears when the Services/Features field is **outwats-bnd** and when ISDN-PRI or ISDN-BRI Trunks field is **y** on the System-Parameters Customer-Options screen. Band is required by Call-by-Call Service Selection.

No. Dgts Subaddress

Allows a caller to reach a number where switch digit processing deletes the dialed number and inserts the listed directory number (LDN). The LDN then is sent to the destination address and the dialed extension is sent in the calling party subaddress information element (IE). At the receiving end, the call terminates to the user indicated by the subaddress number instead of the attendant. Administrable when, on the System-Parameters Customer-Option screen, the ISDN Feature Plus field is **y**.

Valid entries	Usage
1 - 5	Enter the number of dialed digits to send in the calling party subaddress IE.
blank	

Numbering Format

This field applies only to ISDN trunk groups. Enter a value from table below. This field specifies the format of the routing number used for the trunk group for this preference.

This field appears when the ISDN-PRI or ISDN-BRI Trunks field is **y** on the System-Parameters Customer-Options screen.

Valid entries	Numbering Plan Identifier	Type of Numbering
blank	E.164(1)	1-MAX
natl-pub	E.164(1)	national(2)
intl-pub	E.164(1)	international(1)
locl-pub	E.164(1)	local/subscriber(4)
pub-unk	E.164(1)	unknown(0)
lev0-pvt	Private Numbering Plan - PNP(9)	local(4)

Valid entries	Numbering Plan Identifier	Type of Numbering
levl0-pvt (enter to allow Network Call Redirection/ Transfer		
lev1-pvt	Private Numbering Plan - PNP(9)	Regional Level 1(2)
lev2-pvt	Private Numbering Plan - PNP(9)	Regional Level 2(1)
unk-unk	unknown(0)	unknown(0)

NOTE:
To access Bellcore NI-2 Operator Service Access, the Inserted Digits field must be **unk-unk**.

LAR

Enter the routing-preference for Look Ahead Routing.

Valid entries	Usage
next	Go to the next routing preference and attempt the call again.
rehu	Rehunt within the current routing-preference for another trunk to attempt the call again.
none	Look Ahead Routing is not enabled for the preference.

Security-Related System Parameters

You use this screen to determine when the switch reports a security violation. Many of the fields on this screen repeat for each type of security violation. We have explained them once here, but the usage is the same for all. Refer to *“Security violations notification’’ on page 1693* for more information on security violations notification.

Field descriptions for page 1

```
change system-parameters security                              Page 1 of 2
                 SECURITY-RELATED SYSTEM PARAMETERS

SECURITY VIOLATION NOTIFICATION PARAMETERS

  SVN Login Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
              Login Threshold: 5_          Time Interval: 0:03
       Announcement Extension: _____

  SVN Remote Access Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
       Barrier Code Threshold: 10          Time Interval: 0:03
       Announcement Extension: _____

  SVN Authorization Code Violation Notification Enabled? y
        Originating Extension: _____ Referral Destination: _____
 Authorization Code Threshold: 10          Time Interval: 0:03
       Announcement Extension: _____
```

Screen 225. Security-Related System Parameters

SVN Login (Remote Access, AutoStation Security Code, Violation Notification Enabled

Valid entries	Usage
y/n	Set to **y** if you want the switch to notify you when a login violation occurs. If this field is **y**, the next 5 fields appear so you can establish the parameters for what is considered a security violation.

Originating Extension

The originating extension initiates the referral call in the event of a security violation. It also sends the appropriate alerting message or display to the referral destination.

Valid entries	Usage
An unassigned extension	If you establish notification for more than one type of security violations, you must assign a different extension to each one. When the switch generations a referral call, this extension and the type of violation appear on the display at the referral destination.

Referral Destination

The referral destination receives the referral call when a security violation occurs. The referral destination telephone must have a display, unless the you assign an Announcement Extension.

Valid entries	Usage
An extension	Enter the extension of the telephone, attendant console, or vector directory number (VDN) that you want to receive the referral call for each type of violation. This can be the same extension for all type of violations. If you use a VDN, you must complete the Announcement Extension field. You can also use Call Vectoring Time-of-Day routing to route the referral call to different destinations based on the time of day or the day of the week.

Login (Barrier Code, Authorization Code Security Code) Threshold

The value assigned to this field, in conjunction with Time Interval, determines whether a security violation has occurred. For example, if this field is 5, and time interval is 0:03, then five invalid access attempts within three minutes constitutes a security violation.

Valid entries	Usage
1–99, 1–255 for Station Security code	Enter the number of access attempts that are permitted before a referral call is made. In general, it is good to keep this number low. If you are doing testing and do not want to generate alarms, you might change this threshold temporarily.

Time Interval

The value of this field, in conjunction with Threshold, determines whether a security violation has occurred.

Valid entries	Usage
0:01 to **7:59**	The range for the time interval is one minute to eight hours. Entered in the screen “x:xx.” For example, if you want the time interval to be one minute, you enter 0:01. If you want the time interval to be seven and one-half hours, you enter 7:30.

Announcement Extension

If you enter a value in this field, the switch calls the referral destination, then plays this announcement upon answer.

Valid entries	Usage
Valid extension	The announcement extension where SVN violation announcement resides.

SVN Remote Access Violation Notification Enabled

Use these fields to establish parameters for remote access security violations. A remote access violation occurs if a user enters incorrect barrier codes. You cannot set the system to disable remote access following a security violation unless you have turned this on here.

SVN Authorization Code Violation Notification Enabled

Field descriptions for page 2

```
change system-parameters security                                Page 2 of 2
                     SECURITY-RELATED SYSTEM PARAMETERS

  SECURITY VIOLATION NOTIFICATION PARAMETERS

    SVN Station Security Code Violation Notification Enabled? y
            Originating Extension: _____      Referral Destination: _____
  Station Security Code Threshold: 10                Time Interval: 0:03
           Announcement Extension: _____

  STATION SECURITY CODE VERIFICATION PARAMETERS

                        Minimum Station Security Code Length: 4
    Security Code for Terminal Self Administration Required? y

  ACCESS SECURITY GATEWAY PARAMETERS

    SYSAM-LCL? n      SYSAM-RMT? n
        MAINT? n       SYS-PORT? n
  Translation-ID Number Mismatch Interval (days): _
```

Screen 226. Security-Related System Parameters for DEFINITY Server R

```
change system-parameters security                                  Page 2 of 2
                     SECURITY-RELATED SYSTEM PARAMETERS

  SECURITY VIOLATION NOTIFICATION PARAMETERS

    SVN Station Security Code Violation Notification Enabled? y
            Originating Extension: _____      Referral Destination: _____
  Station Security Code Threshold: 10              Time Interval: 0:03
           Announcement Extension: _____

  STATION SECURITY CODE VERIFICATION PARAMETERS

                       Minimum Station Security Code Length: 4
    Security Code for Terminal Self Administration Required? y

  ACCESS SECURITY GATEWAY PARAMETERS

        MGR1? n    INADS? n
         EPN? n      NET? n
```

Screen 227. Security-Related System Parameters for DEFINITY Server SI and DEFINITY Server CSI

SVN Station Security Code Violation Notification Enabled

Station Security codes are used to validate logins to a particular extension (for example, a home agent using an extender, or two part-time workers using the same telephone, but different extensions, through personal station access.) Enter **y** here to establish parameters for this.

Minimum Station Security Code Length

This determines the minimum required length of the Station Security Codes that you enter on the Station screen.

Valid entries	Usage
3–8	Longer codes are more secure. If station security codes are used for external access to telecommuting features, the minimum length should be 7 or 8.

Security Code for Terminal Self Administration Required

Valid entries	Usage
y/n	Enter **y** to indicate that a security code is required.

Access Security Gateway Parameters

These eight fields appear only if the Access Security Gateway (ASG) field on the System-Parameters Customer-Options screen is **y**.

The following 4 fields display only for the DEFINITY Server R version:

SYSAM-LCL

A direct cable connection to the SYSAM-LCL G3-MT (local system administrator's terminal) port on the System Access and Maintenance circuit pack on the active DEFINITY Server R processor carrier. For more information on the circuit pack, refer to the *Avaya MultiVantage™ Hardware Solutions Guide*.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

SYSAM-RMT

A dialed-in (or out) connection to the SYSAM-RMT port on the System Access and Maintenance circuit pack in the active DEFINITY Server R processor carrier. For more information on the circuit pack, refer to the *Avaya MultiVantage™ Hardware Solutions Guide*.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

MAINT

A direct connection to the EPN maintenance circuit pack RS-232 interface.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

SYS-PORT

A dialed-in (or out) connection to the System Access Port.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

The following fields display only for the DEFINITY Server SI and DEFINITY Server CSI versions:

MGR1

The direct connect system administration and maintenance access interface located on the processor circuit pack. For more information on the circuit pack, refer to the *Avaya MultiVantage™ Hardware Solutions Guide*.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

INADS

A direct cable connection to the Initialization and Administration System used to remotely initialize and administer Avaya MultiVantage.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

EPN

A direct connection to the Expansion Port Network.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

NET

A dialed-in (or out) connection to the Network Controller circuit pack. For more information on the circuit pack, refer to the *Avaya MultiVantage™ Hardware Solutions Guide*.

Valid entries	Usage
y/n	Any entry attempt through this port receives a challenge response.

Translation-ID Number Mismatch Interval (days)

A display-only field for all logins except init; only init logins can change this field. This field specifies the interval (in days) that the system allows a mismatch between the translation ID stored in the Processor circuit pack(s) and on the flash card. Following expiration of this interval, the ability to execute system administration commands that modify translation data is denied for all logins, except for init.

Valid entries	Usage
1-90	Enter a number to indicate the number of days the system allows access to system administration commands.

Signaling Group

Use this screen to establish signaling group parameters for ISDN-PRI, H.323, and ATM trunks. Because these trunk types vary in the types of parameters needed, the fields that appear on this screen change depending on the value of the Group Type field

Field descriptions for page 1

```
                                                                 Page 1 of 5
                                 SIGNALING GROUP

Group Number  ___             Group Type:  atm___       Name:
                  Associated Signaling?           Max Number of NCA TSC: ___
                             D-Channel:         Max number of CA TSC:  ___
                                             Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  ___
        Supplementary Service Protocol: _      Network Call Transfer? n
CIRCUIT PARAMETERS
     Virtual Path Identifier: 0
  Virtual Channel Identifier: 0

              Signaling Mode: isdn-pri           Circuit Type: T1
                   Idle Code: 11111111                Connect: network
        Interface Companding: mulaw
            Country Protocol: 1
            Protocol Version: d
                                  DCP/Analog Bearer Capability: 3.1kHz
                                          Interworking Message: PROGress
```

Screen 228. Signaling Group

Group Number

Identifies the signaling group.

Valid entries	Usage
	Identifies the signaling group.

Group type

Valid entries	Usage
Character string	Describe the signaling group.

Name

Appears when the Group Type field is **atm**.

Valid entries	Usage
	Enter alphanumeric characters for identification.

Remote Office

Appears when the Group Type field is **h.323** or **isdn-pri**.

Valid entries	Usage
y/n	Enter **y** if the signaling group serves a remote office.

D Channel

Appears when the Group Type field is **atm**.

Valid entries	Usage
	Enter cabinet number (1-44), carrier (A-E), slot (00-20), circuit (09-32).

Primary D Channel

Appears when the Group Type field is **isdn-pri**.

Valid entries	Usage
	Enter cabinet number (1-44), carrier (A-E), slot (00-20), circuit (09-32).

Max number of NCA TSC

Valid entries	Usage
0-256	Maximum number of simultaneous non-call-associated Temporary Signaling Connections. The TSCs carry signaling for features not associated with a specific call, for example, signals to turn on Leave Word Calling.

Max number of CA TSC

Valid entries	Usage
0-619	Maximum number of simultaneous call-associated Temporary Signaling Connections that can exist in the signaling group. Typically this is the number of ISDN-PRI trunk group members controlled by this signaling group.

Trunk Group for NCA TSC

Valid entries	Usage
1-666 or blank	Enter the ISDN-PRI trunk group number whose incoming call-handling table will be used to handle incoming NCA-TSCs through this signaling group.

Trunk Group for Channel Selection

Valid entries	Usage
1-666 or blank	Enter the trunk group number. If there is more than one trunk group assigned to this signaling group, the group entered in this field will be the one that can accept incoming calls.

Network Call Transfer

Appears only if, on the System-Parameters Customer-Options screen, the ISDN Network Call Redirection field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate D-channels are supporting ENCT.

Supplementary Service Protocol

Valid entries	Usage
a	AT&T, Bellcore, Nortel. When the Country Code field on the DS1 screen is 1A, SSA selects AT&T custom supplementary services. When the Country Code field on the DS1 screen is 1B, SSA selects Bellcore Supplementary Services. When the Country Code field on the DS1 screen is 1C, SSA selects Nortel Proprietary Supplementary Services.
b	ISO QSIG
c	ETSI
d	ECMA QSIG
e	Allows DCS with rerouting. DCS with Rerouting must be **y**, and the Used for DCS field on the trunk group screen must be **y**.
f	Feature Plus
g	ANSI. Available only if, on the System-Parameters Customer-Options screen, the ISDN-PRI or ISDN-BRI field is **y** or the Used for DCS field is **y**.

Network Call Transfer

Appears when the Group Type field is **atm**. Enter **y** in this field for Network Call Transfer. For more information, see *Avaya MultiVantage Network Call Redirection* (555-233-759).

Near-end Node Name

Enter the node name for the C-LAN IP interface on this switch. The node name is administered on the Node Names screen and the IP Interfaces screen.

Valid entries	Usage
Character string	Describe the near-end node.

Far-end Node Name

Enter the node name for the far-end C-LAN IP interface used for trunks assigned to this signaling group. The node name is administered on the Node Names screen.

Valid entries	Usage
Match the Node Names screen.	Describe the far-end node.

Virtual Path Identifier

A display-only field that appears when the Group Type field is **atm**. This field always sets to **0**.

Valid entries	Usage
32-1023 or blank	Enter a number between 32-1023 or blank.

Virtual Channel

Appears when the Group Type field is **atm**.

Valid entries	Usage
32-1023 or blank	Enter a number between 32-1023 or blank.

Near-end Listen Port

Appears when the Group Type field is **h.323**.

Valid entries	Usage
1719, **1720**, or **5000-9999**	Enter an unused port number. Avaya recommends 1720. Enter **1719** if LRQ is **y**.

Far-end Listen Port

Appears when the Group Type field is **h.323**.

Valid entries	Usage
1-65535 or blank	Enter the same number as entered in the Near-end Listen port field.

Far-end Network Region

Appears when the Group Type field is **h.323**. The number of the network region number that is assigned to the far-end of the trunk group.

Valid entries	Usage
1-250 or blank	Enter the network region number that is assigned to the far end of the trunk group. The region is used to obtain the codec set used for negotiation of trunk bearer capability. Leave blank to select the region of the near-end node.

LRQ Required

Appears when the Group Type field is **h.323**.

Valid entries	Usage
y/n	Enter **n** if the far-end switch is an Avaya MultiVantage. Enter **y** if the far-end switch is a non-DEFINITY and requires a location request to obtain a signaling address in its signaling protocol. If **y**, “Calls Share IP Signaling Connection” must be **n**.

Signaling Mode

A display-only field that appears when the Group Type field is **atm**. This field always sets to **isdn-pri**.

Valid entries	Usage
y/n	Enter **n** if the far-end switch is an Avaya MultiVantage. Enter **y** if the far-end switch is a non-MultiVantage and requires a location request to obtain a signaling address in its signaling protocol. If **y**, “Calls Share IP Signaling Connection” must be **n**.

Circuit Type

Appears when the Group Type field is **atm**.

Valid display	Usage
T1	
E1	

RRQ Required

Appears when the Group Type field is **h.323**.

Valid display	Usage
y/n	Displays **y** if the signaling group serves a remote office (gateway). Displays **n** if the signaling group serves a gatekeeper.

Idle Code

Appears when the Group Type field is **atm**.

Valid entries	Usage
0, 1	Enter an 8-digit string.

Connect

Appears when the Group Type field is **atm**.

Valid entries	Usage
host	
network	
pbx	

Calls Share IP Signaling Connection

Appears when the Group Type field is **h.323**.

Valid entries	Usage
y/n	Enter **y** for inter-MultiVantage connections. If **y**, then “LRQ Required” must be n. Enter **n** if the local and/or remote switch is a non-MultiVantage switch.

Interface Companding

Appears when the Group Type field is **atm**.

Valid entries	Usage
alaw	
mulaw	

Bypass If IP Threshold Exceeded

Appears when the Group Type field is **h.323**.

Valid entries	Usage
y/n	Enter **y** to automatically remove from service trunks assigned to this signaling group when IP transport performance falls below limits administered on the Maintenance-Related System Parameters screen.

Country Protocol

Appears when the Group Type field is **atm**.

Valid entries	Usage
1-25	
etsi	

Protocol Version

Appears when the Group Type field is **atm**.

Valid entries	Usage
a	
b	
c	
d	

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter to **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack on the switch.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack on the switch in IP format, without going through the MultiVantage TDM bus.

DCP/Analog Bearer Capability

Appears when the Group Type field is **atm**.

Valid entries	Usage
3.1kHz	
speech	

Interworking Message

Appears when the Group Type field is **atm** or **h.323**. This field determines what message the switch sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group).

Valid entries	Usage
PROGress	Normally select this value. PROGress asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk.
ALERTing	ALERTing causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

Signaling Group screen (page 2)

```
                                                              Page 2 of 5
                        ADMINISTERED NCA TSC ASSIGNMENT
Service/Feature: ___________             As-needed Inactivity Time-out (min):_
  TSC    Local                                                  Adj.      Mach.
 Index    Ext.   Enabled  Established   Dest. Digits    Appl.    Name       ID
  1:     _____     __      _________    _____________   _______  _______    __
  2:     _____     __      _________    _____________   _______  _______    __
  3:     _____     __      _________    _____________   _______  _______    __
  4:     _____     __      _________    _____________   _______  _______    __
  5:     _____     __      _________    _____________   _______  _______    __
  6:     _____     __      _________    _____________   _______  _______    __
  7:     _____     __      _________    _____________   _______  _______    __
  8:     _____     __      _________    _____________   _______  _______    __
  9:     _____     __      _________    _____________   _______  _______    __
 10:     _____     __      _________    _____________   _______  _______    __
 11:     _____     __      _________    _____________   _______  _______    __
 12:     _____     __      _________    _____________   _______  _______    __
 13:     _____     __      _________    _____________   _______  _______    __
 14:     _____     __      _________    _____________   _______  _______    __
 15:     _____     __      _________    _____________   _______  _______    __
```

Screen 229. Signaling Group screen (Administered NCA-TSC Assignment Page)

Field descriptions for page 2

Local Ext

Valid entries	Usage
	Enter the extension of the ISDN interface.

Enabled

Valid entries	Usage
y/n	Enter **y** to enable the administered NCA-TSC.

Established

Used to indicate the strategy for establishing this administered NCA-TSC.

Valid entries	Usage
permanent	Use permanent so that the administered NCA-TSC can be established by either the near end or the far end.
as-needed	Use as-needed so that the administered NCA-TSC will be established the first time the administered NCA-TSC is needed; it can be set up either by the near end or far end switch.

Dest. Digits

Valid entries	Usage
	Enter the extension of the ISDN interface.

Appl.

Specifies the application for this administered NCA-TSC.

Valid entries	Usage
audix	Use this for ISDN-PRI D-channel DCS Audix feature.
dcs	Use this for the DCS Over ISDN-PRI D-channel feature.
gateway	Use this when the administered NCA-TSC is used as one end in the gateway channel connecting to a BX.25 link. If gateway is entered, then the ISDN TSC Gateway Channel Assignments screen must be completed.
masi	Use this when the NCA-TSC is one end of a multimedia application server interface.
qsig-mwi	Use this to convert messages from an administered AUDIX NCA-TSC to a QSIG CISC. If you use this application type, then you must enter a Machine ID between 1 and 20.

Mach ID

You can enter up to 20 machine IDs.

Valid entries	Usage
1 - 20	Enter a unique machine ID. The system does not allow you to specify an ID that you already entered on the Processor Channel screen.

Site Data

Use this screen to enter information about buildings, floors and telephone set colors. You must supply values on this screen before you can enter information in the Site Data section of the Station screen.

```
change site-data
                                SITE DATA USER DEFINITION
                                  VALID BUILDING FIELDS


    _________    _________    _________    _________    _________
    _________    _________    _________    _________    _________
    _________    _________    _________    _________    _________
    _________    _________    _________    _________    _________
    _________    _________    _________    _________    _________
```

Screen 230. Site Data screen

Field descriptions for pages 1, 2 and 3

These pages are available for you to enter descriptive information about the buildings, floors and phone set colors. You may enter any valid keyboard character. If you want to indicate that a particular floor is in a particular building, you must include this in the floor entry, for example, B301-Fl4.

Station

This section provides descriptions of all of the fields that may appear on Station screens. Some of the fields are used for specific phone types; others are used for all phone types. To make it easier to find a specific field, they are listed in alphabetical order by field name.

```
change station 1014                                                Page 1 of X
                              STATION

Extension: 1014               Lock Messages? n                      BCC: 0
     Type:                    Security Code:                         TN: 1
     Port:                  Coverage Path 1:                        COR: 1
     Name:                  Coverage Path 2:                        COS: 1
                            Hunt-to Station:

STATION OPTIONS
              Loss Group: 2              Personalized Ringing Pattern: 3
             Data Module? n                          Message Lamp Ext: 1014
            Speakerphone: 2-way                   Mute button enabled? y
        Display Language? English                    Expansion Module?

                                                   Media Complex Ext:
                                                        IP Softphone? y
```

Screen 231. Station

```
change station 1014                                                Page 1 of X
                                   STATION

Extension: 30016                     Lock Messages? n                   BCC: 0
     Type: XMOBILE                   Security Code:                      TN: 1
                                   Coverage Path 1:                     COR: 1
     Name:                         Coverage Path 2:


STATION OPTIONS
              XMOBILE Type: EC500
      Mobility Trunk Group:                           Message Lamp Ext: 30016
            Display Module? n                     Message Waiting Type: NONE

         Configuration Set: 1

  CELL PHONE NUMBER MAPPING
          Dial Prefix:
    Cell Phone Number:
         Mapping Mode: termination
```

Screen 232. Station when Type field is XMOBILE (page 1)

```
change station 75001                                          Page 2 of X
                                    STATION

FEATURE OPTIONS
           LWC Reception? msa-spe            Auto Select Any Idle Appearance? n
          LWC Activation? y                           Coverage Msg Retrieval? y
  LWC Log External Calls? n                                     Auto Answer: none
             CDR Privacy? n                                 Data Restriction? n
   Redirect Notification? y                     Idle Appearance Preference? n
 Per Button Ring Control? n
   Bridged Call Alerting? n                       Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n            Per Station CPN - Send Calling Number? _
       Service Link Mode: as-needed
         Multimedia Mode: basic
    MWI Served User Type: ______                Display Client Redirection? n
         Automatic Moves:
              AUDIX Name:                      Select Last Used Appearance? n
   Messaging Server Name: ______                 Coverage After Forwarding? _
     Recall Rotary Digit? n                        Multimedia Early Answer? n
      IP Emergency Calls: extension          Direct IP-IP Audio Connections? n
  Emergency Location Ext: 75001                        IP Audio Hairpinning? n
```

Screen 233. Station

```
add station 1014                                                  Page 3 of X
                                    STATION

     SITE DATA
          Room: _______                                   Headset? n
          Jack: ___                                       Speaker? n
         Cable: ___                                      Mounting: d
         Floor: _______                              Cord Length: 0_
      Building: _______                                Set Color: _____


ABBREVIATED DIALING
     List1: ________          List2: _________          List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                4:
 2: call-appr                5:
 3: call-appr
```

Screen 234. Station

```
change station 2014                                              Page 4 0f 4
                                   STATION

FEATURE BUTTON ASSIGNMENTS

                                      2:
                                      3:
 1:                                   4:

 5:                                  12:
 6:                                  13:
 7:                                  14:
 8:                                  15:
 9:                                  16:
10:                                  17:
11:                                  18:
```

Screen 235. Station

If the Expansion Module field is **y**, a fifth page appears.

```
 add station 4005                                              Page 5 of 5
                                      STATION
 EXPANSION MODULE BUTTON ASSIGNMENTS

 1:                                    13:
 2:                                    14:
 3:                                    15:
 4:                                    16:
 5:                                    17:
 6:                                    18:
 7:                                    19:
 8:                                    20:
 9:                                    21:
 10:                                   22:
 11:                                   23:
 12:                                   24:
```

Screen 236. Station screen

The standard station fields are organized alphabetically for easy access.

1-Step Clearing

Valid entries	Usage
y/n	If set to **y**, the call terminates again at the WCBRI terminal when the user drops from the call.

Abbreviated Dialing List1, List2, List3

You can assign up to 3 abbreviated dialing lists to each phone.

Valid entries	Usage
enhanced	Allows the phone user to access the enhanced system abbreviated dialing list.
group	Allows the phone user to access the specified group abbreviated dialing list. If you enter **group**, you also must enter a group number.
personal	Allows the phone user to access and program their personal abbreviated dialing list. If you enter **personal**, you also must enter a personal list number.
system	Allows the phone user to access the system abbreviated dialing list.

Active Station Ringing

Defines how call rings to the phone when it is off-hook. This field does not affect how calls ring at this phone when the phone is on-hook.

Valid entries	Usage
continuous	Enter **continuous** to cause all calls to this phone to ring continuously.
single	Enter **single** to cause calls to this phone to receive one ring cycle and then ring silently.
if-busy-single	Enter **if-busy-single** to cause calls to this phone to ring continuously when the phone is off-hook and idle and calls to this phone to receive one ring cycle and then ring silently when the phone is off-hook and active.
silent	Enter **silent** to cause all calls to this station to just ring silently.

Adjunct Supervision

Adjunct Supervision appears when the Type field is **500**, **2500**, **k2500**, **8110**, **ops**, **ds1fd**, **ds1sa**, **VRU**, **VRUFD**, or **VRUSA**.

Valid entries	Usage
y	Enter **y** if an analog disconnect signal is sent automatically to the port after a call terminates. Analog devices (such as answering machines and speakerphones) use this signal to turn the devices off after a call terminates.
n	Set this field to **n** so hunt group agents are alerted to incoming calls. In a hunt group environment, the disconnect signal blocks the reception of zip tone and incoming call notification by an auto-answer station when a call is queued for the station.

Assigned Member — Ext

The system automatically assigns this extension. This is the extension of the user who has an associated Data Extension button and shares the module.

Assigned Member — Name

Display-only field that shows the name associated with the extension shown in the Assigned Member - Ext field.

Att. Call Waiting Indication

Attendant call waiting allows attendant-originated or attendant-extended calls to a busy single-line phone to wait and sends distinctive call-waiting tone to the single-line user.

Valid entries	Usage
y/n	Enter **y** to assign Attendant Call Waiting to the phone. You should not set this field to y, if the Data Restriction field is **y** or the Switchhook Flash field is **n**, or if Data Privacy is enabled for the phone’s class of service (COS). If any of these conditions are true, the phone cannot accept or handle call waiting calls.

Audible Message Waiting

The Audible Message Waiting tone indicates that the user has a waiting message. This field appears only if Audible Message Waiting is set to **y** on the System-Parameters Customer-Options screen.

Note that this field does not control the Message Waiting lamp.

Valid entries	Usage
y/n	Enter **y** if you want the phone user to receive stutter dial tone when they have a waiting message and they go off-hook.

Audix Name

Specifies which AUDIX is associated with the station.

Valid entries	Usage
Names assigned to an AUDIX adjunct	Must contain a user-defined adjunct name that was previously administered in the Node-Names screen.

Auto Answer

In EAS environments, the auto answer setting on the Agent LoginID screen can override a station’s setting when an agent logs in there.

NOTE:
For analog stations, if Auto Answer is set to acd and the station is off-hook and idle, only the ACD split/skill calls and direct agent calls auto answer; non-ACD calls receive busy treatment. If the station is active on an ACD call and a non-ACD call arrives, the Agent receives call-waiting tone.

Valid entries	Usage
all	Enter all to allow all calls (ACD and non-ACD) terminated to an idle station to be cut through immediately. Does not allow automatic hands-free answer for intercom calls.
acd	Enter acd to allow only ACD split /skill calls and direct agent calls to auto answer. If this field is set to acd, Non-ACD calls terminated to a station ring audibly.
none	Enter none to cause all calls terminated to this station to receive an audible ringing treatment.
icom	Enter icom to allow a phone user to answer an intercom call from the same intercom group without pressing the intercom button.

Automatic Moves

Automatic Moves allows a phone to be unplugged from one location and moved to a new location without additional switch administration. The switch automatically associates the extension to the new port.

CAUTION:

When a phone is unplugged and moved to another physical location, the Emergency Location Extension field must be changed for that extension or the USA Automatic Location Identification data base must be manually updated. If the Emergency Location Extension field is not changed or if the USA Automatic Location Identification data base is not updated, the DID number sent to the Public Safety Network could send emergency response personnel to the wrong location.

Valid entries	Usage
always	Enter always and the phone can be moved anytime without additional administration by unplugging from one location and plugging into a new location.
once	Enter once and the phone can be unplugged and plugged into a new location once. After a move, the switch sets the field to done the next time routine maintenance runs on the phone. Use once when moving a large number of phones so each extension is removed from the move list. Use once to prevent automatic maintenance replacement.
no	Enter no to require administration in order to move the phone.
done	Done is a display-only value. The switch sets the field to done after the phone is moved and routine maintenance runs on the phone.
error	Error is a display-only value. The switch sets the field to error, after routine maintenance runs on the phone, when a non-serialized phone is set as a movable phone.

Auto Select Any Idle Appearance

Valid entries	Usage
y/n	Enter **y** to allow automatic selection of any idle appearance for transferred or conferenced calls. The system first attempts to find an idle appearance of the call being transferred or conferenced. If that attempt fails, the system selects the first idle appearance.

Automatic Selection of DID Numbers

The switch chooses a 2- to 5-digit extension from a predetermined list of numbers and assigns the extension to a hotel room phone.

Valid entries	Usage
y/n	Enter **y** to use the Automatic Selection of DID Numbers for Guest Rooms feature.

BCC

Appears when ISDN-PRI or ISDN-BRI Trunks is enabled on the System-Parameters Customer-Options screen. Display-only field set to **0** for stations (that is, indicates voice or voice-grade data).

Refer to ''Generalized route selection'' on page 1567 for a detailed description of Bearer Capability Classes (BCC) and their ability to provide specialized routing for various types of voice and data calls. The BCC value is used to determine compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking).

Bridged Call Alerting

If Bridged Call Alerting is **n** and Per Button Ring Control is **n**, audible ringing is suppressed for incoming calls on bridged appearances of another phone's primary extension.

Valid entries	Usage
y/n	Enter **y** to enable audible ringing for TEG, PCOL, bridged appearances, or Data Extension calls.

Building

Enter a valid building location. See *''Signaling Group screen (page 2)'' on page 1063* for valid entries.

Busy Auto Callback Without Flash

Appears only if, on the Feature-Related System Parameters screen, the Without Flash field is **y**. This field then defaults to **y** for all analog phones that allow Analog Automatic Callback.

Valid entries	Usage
y/n	Enter **y** to provide automatic callback for a calling analog station without flashing the hook.

Button Assignments

Enter the abbreviated software name to assign a feature button. For a list of feature buttons, refer to ''Telephone feature buttons'' on page 129.

NOTE:
If you want to use Terminal Translation Initialization (TTI), you must assign a call appearance (call-appr) to the first button position. TTI needs the button on the first call appearance to get dial tone.

Cable

You can use this field to identify the cable that connects the phone jack to the system. You also can enter this information in the Blank column on the Port Assignment Record.

Caller ID Message Waiting Indication

Appears when the Type field is **CallrID**. For CallrID type phones or analog phones with Caller ID adjuncts only.

Valid entries	Usage
y/n	Enter **y** to allow aliasing of various non-Avaya phones and adjuncts.

NOTE:
The Caller ID Message Waiting Indication administration is independent of the administration of LED or NEON lamp Avaya MultiVantage Message Waiting Indication (MWI). For example, it is possible to administer a Caller ID phone with the Caller ID Message Waiting Indication field set to **n** and the Message Waiting Indicator field set to **neon**.

Call Waiting Indication

This allows user, attendant-originated, and outside calls to busy single-line phone to wait and sends a distinctive call-waiting tone to the single-line user. This feature is denied if Data Restriction is **y** or Switchhook Flash is **n**, or if Data Privacy is active via the phone COS assignment.

Valid entries	Usage
y/n	Enter **y** to activate Call Waiting Termination for the phone.

CDR Privacy

This option allows digits in the called number field of an outgoing call record to be blanked, on a per-station basis. You administer the number of blocked digits system-wide in the Privacy - Digits to Hide field on the CDR System Parameters screen.

Valid entries	Usage
y/n	Enter **y** to enable Call Privacy for each station.

Cell Phone Number

Contains the unformatted cell phone's published external number. This field may contain a 3-digit area code plus the 7-digit main number. If the same Cell Phone Number is administered on multiple XMOBILE station screens, then the Dial Prefix associated with each instance of the Cell Phone Number must be the same.

Valid entries	Usage
0-9	Enter 1-15 digits. Avaya recommends that you enter a full 10-digit Cell Phone Number regardless of whether the cell phone is local or not. Note that your ARS screen has to be administered to handle this.

Configuration Set

Valid entries	Usage
1-10	Enter any value corresponding to the appropriate Configuration Set screen.

COR

Enter a Class of Restriction (COR) number to select the desired restriction.

Cord Length

Enter a number to specify the length (in feet) of the cord attached to the receiver.

COS

Enter the desired Class of Service (COS) number to select allowed features.

Country Protocol

Enter the protocol that corresponds to your supported initialization and codesets. The Country Protocol must match any previously-administered endpoint on the same port.

Valid entries	Usage
1	United States (Bellcore National ISDN)
2	Australia
etsi	ETSI (Europe)
3	Japan
6	Singapore

Coverage After Forwarding

This field governs whether an unanswered forwarded call is provided coverage treatment.

Valid entries	Usage
y	Coverage treatment is provided after forwarding regardless of the value of the Coverage After Forwarding field on the System Parameters - Call Coverage/Call Forwarding screen.
n	No coverage treatment is provided after forwarding regardless of the value of the Coverage After Forwarding field on the System Parameters - Call Coverage/Call Forwarding screen.
s(ystem)	Indicates that call processing uses the Coverage After Forwarding field on the ''System Parameters Call Coverage / Call Forwarding'' screen. To override the system-wide parameter for a given station, set this field to **y** or **n**.

Coverage Msg Retrieval

Applies if the phone is enabled for LWC Reception.

Valid entries	Usage
y/n	Enter **y** to allow users in the phone's Coverage Path to retrieve Leave Word Calling (LWC) messages for this phone.

Coverage Module

Valid entries	Usage
y	Enter **y** to indicate that a coverage module is connected to the station. Once you enter y, the system displays an additional page that allows you to assign the buttons for the module.

Coverage Path 1 or Coverage Path 2

Enter a coverage-path number or time-of-day table number from a previously-administered Call Coverage Path screen or Time of Day Coverage Table screen.

NOTE:
If Modified Misoperation is active (Misoperation Alerting is **y** on the Feature-Related System Parameters screen), you must assign a Coverage Path to all stations on the switch.

CRV Length

Only for ASAI stations. Enter **1** or **2** to indicate the length of CRV for each interface.

Custom Selection of VIP DID Numbers

Custom Selection of VIP DID numbers allows you to select the DID number assigned to a room when a guest checks in.

Valid entries	Usage
y/n	Enter **y** to allow you to select the DID number assigned to a room when a guest checks in.

Data Extension

Enter the extension assigned to the data module.

Data Option

Valid entries	Usage
analog	
data module	
none	

Data Restriction

Data restriction provides permanent protection and cannot be changed by the phone user. Do not assign a Data Restriction if Auto Answer is **all** or **acd**. If **y**, whisper page to this station is denied.

Valid entries	Usage
y/n	Enter **y** to prevent tones, such as call-waiting tones, from interrupting data calls.

Default Dialing Abbreviated Dialing Dial Code

Appears only when the Special Dialing Option is set to default. Enter a list number associated with the abbreviated dialing list.

When the user goes off-hook for a data call and presses the Return button following the DIAL prompt, the system dials the AD number. The data call originator also can perform data-terminal dialing by specifying a dial string that may or may not contain alphanumeric names.

Dial Prefix

Contains the unformatted sequence of digits or characters that are prepended to the cell phone's published cell phone number before dialing. If the same Cell Phone Number is administered on multiple XMOBILE station screens, then the Dial Prefix associated with each instance of the Cell Phone Number must be the same.

Valid entries	Usage
0-9, *, #	Enter 1-4 digits.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

Display Caller ID

Appears when the Type field is **CallrID**. For CallrID type phones or analog phones with Caller ID adjuncts only.

Valid entries	Usage
y/n	Enter **y** to allow transmission of calling party information to the Caller ID phone or adjunct.

Display Cartridge

For 7404 D phones only. Enter **y** to indicate there is a display cartridge associated with the station. This displays an additional page to allow you to assign display buttons for the display cartridge.

Display Client Redirection

Only administrable if Hospitality is enabled on the System-Parameters Customer-Options screen. This field affects the phone display on calls that originated from a station with Client Room Class of Service.

NOTE:
For stations with an audix station type, AUDIX Voice Power ports, or ports for any other type of messaging that needs display information, Display Client Redirection must be set to **y**.

Valid entries	Usage
y	When set to **y**, the redirection information for a call originating from a Client Room and terminating to this station displays.
n	When set to **n**, this station's display does not show the redirection information for all calls originating from a Client Room (even redirected calls) that terminate to this station. Only the client name and extension (or room, depending on what is administered on the ''Hospitality'' screen) display.

Display Language

Specifies the display language.

Valid entries	Usage
english	Enter the language you want users to see on their displays.
french	
italian	
spanish	
user-defined	

Display Module

Valid entries	Usage
y/n	Enter **y** if this phone has a display module.

Distinctive Audible Alert

Valid entries	Usage
y/n	Enter **y** so the phone can receive the 3 different types of ringing patterns which identify the type of incoming calls. Distinctive ringing may not work properly for off-premises telephones.

Emergency Location Ext

The Emergency Location Ext field defaults to the phone's extension. This extension identifies the street address or nearby location when an emergency call is made.

Valid entries	Usage
1-8 digits	Enter the Emergency Location Extension for this station

Endpt ID

Appears only if Endpt Init is **y**. Enter a unique 2-digit number (**00**–**62**) for this endpoint. Each Endpt ID field must have a unique value for each endpoint on the same port.

This field provides for multipoint configuration conformance to the Bellcore Terminal Initialization procedures. In these procedures, a multipoint configuration requires the last 2 digits of the Service Profile Identifier (SPID) be between 00 and 63 and be binary unique for each endpoint. This field, combined with the SPID, gives the effective SPID administered into the terminal. Bellcore ISDN-1 requires the SPID programmed into the endpoint contain at least 9 digits.

For example, if the SPID is **1234**, and Endpt ID is **01**, then the SPID administered on the terminal is 000123401. The three leading zeros are necessary to create a 9-digit SPID.

Endpt Init

Endpoint initialization is a procedure, required for multipoint operation, by which User Service Order Profile (USOP) is associated with an endpoint on the ISDN-BRI. This association is made via the SPID, administered into the system, and entered into the ISDN-BRI terminal. For an ISDN-BRI terminal to be operational in a multipoint configuration, both the administered SPID and the SPID programmed into the ISDN-BRI terminal must be the same. Therefore, the SPID of new or reused terminals must be programmed to match the administered SPID value. Appears only if MIM Support is **y** and indicates the terminal’s endpoint initialization capability.

Valid entries	Usage
y	Enter **y** if the terminal supports Bellcore ISDN-1 terminal initialization procedures.
n	Enter **n** for all other country protocols.

Expansion Module

Valid entries	Usage
y/n	Enter **y** if this phone has an expansion module. This will enable you to administer the buttons for the expansion module.

Extension

Displays the extension for this station—this is either the extension you specified in the station command or the next available extension (if you used add station next).

For a virtual extension, enter a valid physical extension or a blank. Blank allows an incoming call to the virtual extension to be redirected to the virtual extension's "busy" or "all" coverage path.

> We recommend that you consider the display for emergency notification when you complete the name field on the station screen. Put the most important identifying information at the beginning of the field. When an emergency call is made and a crisis alert station with a 27-character display is notified, only 17 characters of the name field appear on the first display line, followed by the extension. The second line contains the last 3 characters of the name field, followed by the word "EMERGENCY." Characters 18 through 24 of the name field do not appear at all.

Feature Module

Enter **y** to indicate the station is connected to a feature module. Entering **y** displays an additional page to allow you to assign feature buttons to the module.

Fixed TEI

This field appears only for ISDN-BRI data modules and ASAI links. It indicates that the endpoint has a fixed Terminal Endpoint Identifier (TEI).

The TEI identifies a unique access point within a service. You must administer TEIs for fixed TEI terminals. However, for terminals with the automatic TEI capability, the system dynamically assigns the TEI.

Valid entries	Usage
y/n	Entering **y** displays the TEI field. For ASAI, enter **y**.

Floor

Enter a valid floor location. See ''Translation-ID Number Mismatch Interval (days)'' on page 1054 for valid entries.

H.320 Conversion

Allows H.320 compliant calls made to this phone to be converted to voice-only. Because the system can only handle a limited number of conversion calls, you may need to limit the number of telephones with H.320 conversion.

Headset

Enter **y** if the telephone has a headset.

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

Appears only when Special Dialing Option is **hot-line.**

Use Hot Line Service when very fast service is required and when you use a telephone only for accessing a certain facility. Hot Line Service allows single-line telephone users, by simply lifting the handset, to automatically place a call to a preassigned destination (extension, telephone number, or feature access code).

The Hot Line Service destination number is stored in an Abbreviated Dialing List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.

A Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension, or any individual extension within a group can be a Hot Line Service destination. Also, any extension within a DDC group, UDC group, or TEG can have Hot Line Service assigned.

Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

HOT LINE DESTINATION — Abbreviated Dialing List Number

Enter the abbreviated dialing list where you stored the hotline destination number.

HOT LINE DESTINATION — Dial Code

Enter the dial code in the specified abbreviated dialing list where you stored the hotline destination number.

Hunt-to Station

Enter the extension the system should hunt to for this phone when the phone is busy. This field allows you to create a station hunting chain (by assigning a hunt-to station to a series of phones).

Idle Appearance Preference

Indicate which call appearance is selected when the user lifts the handset and there is an incoming call.

Valid entries	Usage
y	If you enter **y**, the user connects to an idle call appearance instead of the ringing call.
n	If you enter **n**, the Alerting Appearance Preference is set and the user connects to the ringing call appearance.

Ignore Rotary Digits

If this field is **y**, the short switch hook flash (50 -150) from a 2500-type set is ignored.

Valid entries	Usage
y	Enter **y** to indicate that rotary digits from the set should be ignored.
n	Enter **n** to make sure they are not ignored.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack on the switch.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack on the switch in IP format, without going through the MultiVantage TDM bus.

IP Emergency calls

Use this field to tell the switch how to handle emergency calls from the IP phone. This field appears when either the IP Softphone field or the Remote Office Station field is set to **y** on the Station screen.

CAUTION:

An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It reaches solely the local emergency service in the Public Safety Answering Point area where the telephone system is located. Please be advised that an Avaya IP endpoint does not dial to and connect with local emergency service when dialing from remote locations. You should not use an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not be responsible or liable for any damages resulting from misplaced emergency calls made from *an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations.*

Valid entries	Usage
extension	Enter **extension** to send the extension entered in the Emergency Location Extension field, to the Public Safety Answering Point (PSAP).
block	Enter **block** to prevent the completion of emergency calls. Use this entry for users who move around but always have a circuit-switched phone nearby, and for users who are farther away from the switch than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP Telephone and the call is blocked, they can dial 911 from a nearby circuit-switched phone instead.
cesid	Enter **cesid** to allow the switch to send the CESID information supplied by the IP Softphone to the PSAP. The end user enters the emergency information into the IP Softphone. Use this entry for IP Softphones with road warrior service that are near enough to the switch that an emergency call routed over the switch's trunk reaches the PSAP that covers the switch. If the switch uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP Softphone. If the switch uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP Softphone location, based on advice from the local emergency response personnel.
option	Enter **option** to allow the user to select the option (extension, block, or cesid) that the user selected during registration and the IP Softphone reported. Use this entry for extensions that are swapped back and forth between IP Softphones and a phone with a fixed location. The user chooses between **block** and **cesid** on the softphone. A DCP or IP phone in the office automatically selects **extension**.

IP Softphone

Appears only for DCP station types and IP Telephones.

Valid entries	Usage
y/n	Enter **y** indicate that this extension is either a PC-based multifunction station or part of a telecommuter complex with a call-back audio connection.

ITC (Information Transfer Capability)

Indicates the type of transmission facilities to be used for ISDN calls originated from this endpoint. The field does not display for voice-only or BRI stations.

Valid entries	Usage
restricted	If you set to **restricted**, either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission (that is, a sequence of 8 digital zeros are converted to a sequence of 7 zeros and a digital 1).
unrestricted	If you set to **unrestricted**, only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission (that is, digital information is sent exactly as is).

Lock Messages

Valid entries	Usage
y/n	Enter **y** to restrict other users from reading or canceling the voice messages or retrieving messages via Voice Message Retrieval.

Loss Group

This field determines which administered 2-party row in the loss plan applies to each station. Does not appear for stations that do not use loss (for example, x-mobile stations and MASI terminals).

Valid entries	Usage
1-17	Shows the index into the loss plan and tone plans.

LWC Activation

Valid entries	Usage
y/n	Enter **y** to allow internal telephone users to leave short LWC messages for this extension. If the system has hospitality, enter **y** for guest-room telephones if the extension designated to receive failed wakeup messages should receive LWC messages that indicate the wakeup calls failed. Enter **y** if LWC Reception is **audix**.

LWC Log External Calls

Appears only where the LWC Reception field is available. When an external call is not answered, the switch keeps a record of up to 15 calls (provided information on the caller identification is available) and the phone's message lamp lights. The phone set displays the names and numbers of unsuccessful callers.

Valid entries	Usage
y/n	Enter **y** to make unanswered external call logs available to end-users. Each record consists of the latest call attempt date and time.

LWC Reception

Valid entries	Usage
audix	Enter **audix** if the messages are stored on the Audio Information Exchange System.
none	
msa-spe	Enter **msa-spe** if LWC messages are stored in the system or on the Messaging Server Adjunct - Switch Processor (for DEFINITY Server SI / DEFINITY Server CSI models).
spe	Enter **spe** if LWC messages are stored in the system or on the Switch Processor (for r models).

Mapping Mode

Controls whether an EC500 cell phone is administered to gain the identity of the office phone (sends the office caller ID).

Valid entries	Usage
both	The cell phone can be used for both sending office caller ID and receiving EC500 calls.
none	The XMOBILE station is disabled administratively.
origination	The cell phone may only be used to originate calls from its associated internal XMOBILE extension, by dialing into the office switch.
termination	The cell phone may only be used to terminate calls from its associated internal XMOBILE extension.

Map-to Station

This is the physical phone used for calls to a virtual extension. Do not use an xmobile, xdid or another virtual extension.

Valid entries	Usage
Valid extension number	Enter the extension of the physical phone used for calls to a virtual extension.

Media Complex Ext

When used with Multi-media Call Handling, indicates which extension is assigned to the data module of the multimedia complex. Users can dial this extension to place either a voice or a data call, and voice conversion, coverage, and forwarding apply as if the call were made to the 1-number.

For an IP Telephone or an IP Softphone, this is the extension already administered as an H.323 station type. You can administer this field if the IP Station field on the System-Parameters Customer-Options screen is **y**.

Valid entries	Usage
A valid BRI data extension	For MMCH, enter the extension of the data module that is part of this multimedia complex.
H.323 station extension	For 4600 series IP Telephones, enter the corresponding H.323 station.
	For IP Softphone, enter the corresponding H.323 station. If you enter a value in this field, you can register this station for either a road-warrior or telecommuter/Avaya IP Agent application.
blank	Leave this field blank for single-connect IP applications.

Message Lamp Ext

Enter the extension of the station you want to track with the message waiting lamp. This field appears only when Type is **7101A**, **7103A**, **8110**, or **VRU**.

Message Server Name

Specifies which Message Server is associated with the station.

Valid entries	Usage
Names administered in alphabetical order	Must contain a user-defined adjunct name that was previously administered in the Node-Names screen.

Message Waiting Indicator

This field appears only for ISDN-BRI data modules and for 500, 2500, K2500, 7104A, 6210, 6218, 6220, 8110, and VRU telephones. (This field is independent of the administration of the Caller ID Message Waiting Indication for CallrID phones.)

Valid entries	Usage
led	Enter **led** if the message waiting indicator is a light-emitting diode (LED).
neon	Enter **neon** if the indicator is a neon indicator.

MIM Mtce/Mgt

Indicates if the telephone supports MIM Maintenance and Management capabilities other then endpoint initialization. Appears only if MIM Support is **y**.

MIM Support (Management Information Message Support)

This field appears only for ISDN-BRI data modules and ASAI. This field supports MIM endpoint initialization (SPID support) and other Maintenance or Management capabilities.

Valid entries	Usage
y	Enter **y** to display Endpt Init and MIM Mtce/Mgt.
n	Enter **n** for ASAI.

Multimedia Early Answer

Allows you to establish multimedia early answer on a station-by-station basis.

Valid entries	Usage
y/n	If this station will receive coverage calls for multimedia complexes, but is not multimedia-capable, enter **y** to ensure that calls are converted and talk path is established before ringing at this station.

Mute Button Enabled

Valid entries	Usage
y/n	Enter **y** to allow the user to use the Mute button on this phone.

MWI Served User Type

Controls the auditing or interrogation of a served user's message waiting indicator (MWI).

Valid entries	Usage
fp-mwi	Use if the station is a served user of an fp-mwi message center.
qsig-mwi	Use if the station is a served user of a qsig-mwi message center.
blank	Leave this field blank if you do not want to audit the served user's MWI or if the user is not a served user of either an fp-mwi or qsig-mwi message center.

Name

Enter a name for the person associated with this phone or data module. The system uses the Name field to create the system Directory.

Off Premises Station

Analog phones only.

Valid entries	Usage
y	Enter **y** if this phone is not located in the same building with the system. If you enter **y**, you must complete R Balance Network.
n	Enter **n** if the phone is located in the same building with the system.

PCOL/TEG Call Alerting

Appears only for 510 telephones.

Valid entries	Usage
y/n	Enter **y** to alert the station for Personal CO Line/Terminating Extension Group calls.

Per Button Ring Control

Valid entries	Usage
y	Enter **y** to allow users to select ring behavior individually for each call-appr, brdg-appr, or abrdg-appr on the station and to enable Automatic Abbreviated and Delayed ring transition for each call-appr on the station. Also, enter **y** if you do not want the system to automatically move the line selection to a silently alerting call unless that call was audibly ringing earlier.
n	Enter **n** if you want calls on call-appr buttons to always ring the station and calls on brdg-appr or abrdg-appr buttons to always ring or not ring based on the Bridged Call Alerting field value. Also, enter **n** if you want the system to move line selection to a silently alerting call if there is no call audibly ringing the station.

Personalized Ringing Pattern

Personalized Ringing allows users of some telephones to have one of 8 ringing patterns for incoming calls. Users working closely in the same area can each specify a different ringing pattern. This enables the users to distinguish their own ringing telephone from other telephones in the same area. For virtual stations, this field dictates the ringing pattern on its mapped-to physical phone.

Enter a Personalized Ringing Pattern. (L = 530 Hz, M = 750 Hz, and H = 1060 Hz)

Valid entries	Usage
1	MMM (standard ringing)
2	HHH
3	LLL
4	LHH
5	HHL
6	HLL
7	HLH
8	LHL

Per Station CPN - Send Calling Number

Valid entries	Usage
y	All outgoing calls from the station will deliver the Calling Party Number (CPN) information as “Presentation Allowed.”
n	No CPN information is sent for the call.
r	Outgoing non-DCS network calls from the station will deliver the Calling Party Number information as “Presentation Restricted.”
blank	The sending of CPN information for calls is controlled by any administration on the outgoing trunk group the calls are carried on.

Port

Enter 7 characters to specify a port, or an x. If this extension is registered as an IP endpoint, this field will display S00000.

Valid entries	Usage
01 through **44** (DEFINITY Server R)	First and second numbers are the cabinet number
01 through **03** (DEFINITY Server SI)	
A through **E**	Third character is the carrier
01 through **20**	Fourth and fifth characters are the slot number
01 through **32**	Sixth and seventh characters are the circuit number
x	Indicates that there is no hardware associated with the port assignment. Use for AWOH and CTI stations.
IP	Automatically entered for IP phones.

For DCP sets, the port can only be assigned once. ISDN-BRI provides a multipoint configuration capability that allows a previously assigned port to be specified more than once as follows: 2 stand-alone voice endpoints, 2 stand-alone data endpoints, or 1 integrated voice and data endpoint.

However, for the following cases, the port is assumed to be fully assigned:

- Maximum number of users (currently 2) are assigned on the port.
- One of the users on the port is a fixed TEI station.
- One of the users on the port has B-channel voice and B-channel data capability.
- One of the users on the port has no SPID assigned, which includes telephones that have no SPID initialization capability.
- This field is display-only for H.323 set types and 4600 set types. The system assigns an “IP” when the station is first administered.

R Balance Network

Valid entries	Usage
y	Enter **y** to select the R Balance Capacitor network. In all other cases except for those listed under **n**, enter **y**.
n	Enter **n** to select the standard resistor capacitor network. You must complete this field if Off-Premise Station is **y**. Enter **n** when the station port circuit is connected to terminal equipment (such as SLC carriers or impedance compensators) optioned for 600-ohm input impedance and the distance to the equipment from the system is less than 3,000 feet.

Recall Rotary Digit

This field only appears if type is 500 or 2500.

Valid entries	Usage
y/n	Enter **y** to allow the user of a rotary phone to dial the administered Recall Rotary Digit to receive recall dial tone. This will enable this user to perform conference and transfer operations. You establish the Recall Rotary Digit on the Feature-Related System Parameters screen.

Redirect Notification

Valid entries	Usage
y/n	Enter **y** to give a half ring at this phone when calls to this extension are redirected (via Call Forwarding or Call Coverage). Enter **y** if LWC Reception is **audix**.

Remote Office Phone

Valid entries	Usage
y/n	Enter **y** to use this station as an endpoint in a remote office configuration.

Restrict Last Appearance

Valid entries	Usage
y/n	Enter **y** to restrict the last idle call appearance for incoming priority calls and outgoing call originations only.

Room

Valid entries	Usage
Up to 10 characters	To identify the phone location.
Up to 5 characters	To identify the guest room number, if this station is one of several to be assigned a guest room and the Display Room Information in Call Display is **y** on the Hospitality-Related System Parameters screen.

Security Code

Enter the security code required by users for specific system features and functions, including Personal Station Access, Redirection of Calls Coverage Off-Net, Leave Word Calling, Message Retrieval, and Demand Printing. The required security code length is determined by Minimum Security Code Length on the Feature-Related System Parameters screen.

Select Last Used Appearance

Valid entries	Usage
y	Enter **y** to indicate a station's line selection is not to be moved from the currently selected line button to a different, non-alerting line button. If you enter **y**, the line selection on an on-hook station only moves from the last used line button to a line button with an audibly alerting call. If there are no alerting calls, the line selection remains on the button last used for a call.
n	Enter **n** so the line selection on an on-hook station with no alerting calls can be moved to a different line button, which may be serving a different extension.

Service Link Mode

The service link is the combined hardware and software multimedia connection between an Enhanced mode complex's H.320 DVC system and the Avaya MultiVantage which terminates the H.320 protocol. When the user receives or makes a call during a multimedia or IP Softphone or IP Telephone session, a "service link" is established.

Valid entries	Usage
as-needed	Use this setting for most multimedia, IP Softphone, or IP Telephone users. Setting the Service Link Mode to as-needed leaves the service link connected for 10 seconds after the user ends a call so that they can immediately place or take another call. After 10 seconds the link is dropped and a new link would have to be established to place or take another call.
permanent	Use **permanent** for busy call center agents and other users who are constantly placing or receiving multimedia, IP Softphone, or IP Telephone calls. In permanent mode, the service link stays up for the duration of the multimedia, IP Softphone, or IP Telephone application session.

Set Color

Enter a valid set color as defined in the Signaling Group screen (page 2) screen. Valid entries include the following colors: beige, black, blue, brown, burg (burgundy), gray, green, ivory, orng (orange), red, teak, wal (walnut), white, and yel (yellow).

Speakerphone

This field controls the behavior of speakerphones.

Valid entries	Usage
1-way	Enter **1-way** to indicate that you want the speakerphone to be listen-only.
2-way	Enter **2-way** to indicate that you want the speakerphone to be both talk and listen.
grp-listen	Group Listen works with only 6400-series and 2420 phones. Group Listen allows a phone user to talk and listen to another party with the handset or headset while the phone's 2-way speakerphone is in the listen-only mode. Others in the room can listen, but cannot speak to the other party via the speakerphone. The person talking on the handset acts as the spokesperson for the group. Group Listen provides reduced background noise and improves clarity during a conference call when a group needs to discuss what is being communicated to another party.
none	

Special Dialing Option

This field identifies the type of dialing for calls when this data module originates calls.

Valid entries	Usage
hot-line	
default	
blank	For regular (normal) keyboard dialing.

SPID — (Service Profile Identifier)

Enter a variable length parameter. This field appears only if the Endpt Init field is **y**.

The SPID is a numeric string, which means that the value of 00 is different from 000. The SPID must be different for all terminals on the BRI and from the Service SPID. The SPID should always be assigned. If the SPID is not assigned for the first BRI on a port, any other BRI assignment to that port are blocked.

NOTE:
If you have set the Port field to X for an ISDN-BRI extension and intend to use Terminal Translation Initialization (TTI) to assign the port, then the SPID number must equal the station number.

Station Lock Active

Valid display	Usage
yes/no	Shows the phone’s status of Station Lock.

Switchhook Flash

Valid entries	Usage
y	Enter **y** to allow users to use the switchhook flash function to activate Conference/Transfer/Hold and Call Waiting.
n	Enter **n** to disable the flash function so that when the switchhook is pressed while active on a call, the call drops. If this field is **n**, you must set Call Waiting Indication to **n**.

TEI

Appears only when the Fixed TEI field is **y**.

Valid entries	Usage
0 to **63**	1- or 2-digit number

Tests

Valid entries	Usage
y	Enter **y** to enable port maintenance tests.
n	If the equipment (dictaphone) connected to the port does not support these tests, you must enter **n**.

TN

Enter the Tenant Partition number.

Type

For each station that you want to add to your system, you must specify the type of telephone in the Type field. This is how you distinguish between the many different types of telephones.

The following table lists the telephones, virtual phones, and personal computers that you can administer on the Avaya MultiVantage. To administer telephones that are not in the table, use the Alias Station screen.

NOTE:
You cannot change an analog phone administered with hardware to a virtual extension if TTI is **y** on the Feature-Related System Parameters screen. Contact your Avaya representative for more information.

Table 22. telephones

Telephone type	Model	Administer as
Single-line analog	500	500
	2500, 2500 w/ Message Waiting Adjunct	2500
	6210	6210
	6211	6210
	6218	6218
	6219	6218
	6220	6220
	6221	6220
	Analog phone w/Caller ID	CallrID
	7101A, 7102A	7101A
	7103A Programmable and Original	7103A
	7104A	7104A
	8110	8110
	DS1FD	DS1FD
	7302H, 7303H	7303S
	VRU (voice response unit) with C&D tones	VRU
	VRU without C&D tones	2500
Single-line DS1/DSO (Lineside T1/DS1)	DS1 device without forward disconnect	ops
	VRU with forward disconnect without C&D tones	ds1fd or ds1sa

Continued on next page

Table 22. telephones (Continued)

Telephone type	Model	Administer as
	VRU with forward disconnect without C&D tones	VRUFD or VRUSA
Terminals	510D	510
	515BCT	515
Multiappearance hybrid	7303S	7303S, 7313H
	7305H	7305S
	7305S	7305S, 7316H, 7317H
	7309H	7309H, 7313H
	7313H	7313H
	7314H	7314H
	7315H	7315H
	7316H	7316H
	7317H	7317H
Multiappearance digital	2420	2420
	6402	6402
	6402D	6402D
	6408	6408
	6408+	6408+
	6408D	6408D
	6408D+	6408D+
	6416D+	6416D+
	6424D+	6424D+
	7401D	7401D
	7401+	7401+
	7403D	7403D
	7404D	7404D
	7405D	7405D

Continued on next page

Table 22. telephones (Continued)

Telephone type	Model	Administer as
	7406D	7406D
	7406+	7406+
	7407D	7407D
	7407+	7407+
	7410D	7410D
	7410+	7410+
	7434D	7434D
	7444D	7444D
	8403B	8403B
	8405B	8405B
	8405B+	8405B+
	8405D	8405D
	8405D+	8405D+
	8410B	8410B
	8410D	8410D
	8411B	8411B
	8411D	8411D
	8434D	8434D
	CALLMASTER I	602A1
	CALLMASTER II, III, IV	603A1, 603D1, 603E1, 603F1
	CALLMASTER VI	606A1
	IDT1	7403D
	IDT2	7406D
IP Telephone	4602	4602
	4606	4606
	4612	4612

Continued on next page

Table 22. telephones (Continued)

Telephone type	Model	Administer as
	4620	4620
	4624	4624
	4630	4630
IP SoftPhone	Road-warrior application	H.323 or DCP type
	Native H.323	H.323
	Single-connect	H.323 or DCP type
ISDN-BRI station	—	asai
	7505D	7505D
	7506D	7506D
	7507D	7507D
	8503D	8503D
	8510T	8510T
	8520T	8520T
Personal computer	6300/7300	PC
(voice/data)	6538/9	Constellation
Test Line	ATMS	105TL
No hardware assigned at the time of administration.		XDID (use when the switch later assigns a DID number to this station)
		XDIDVIP (use when the administrator later assigns a DID number to this station)
		virtual (use to map this and other extensions to one physical phone)
Key phone system interface	—	K2500

Continued on next page

Table 22. telephones (Continued)

Telephone type	Model	Administer as
ASAI	asai link	asai
	computer telephony adjunct link	adjlk
AWOH	any digital set	same as “Multiappearance Digital” see table above
	CTI station	CTI
CTI	CTI station	CTI
XMOBILE	EC500, DECT, PHS	XMOBILE
ISDN-BRI data module	7500	7500

XID

Appears only for an ISDN-BRI data module or an ASAI link. Used to identify Layer 2 XID testing capability.

System Parameters Call Coverage / Call Forwarding

This screen sets the system-wide parameters for call coverage and call forwarding.

Field descriptions for page 1

```
 change system-parameters coverage-forwarding (page 1)

    SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

CALL COVERAGE/FORWARDING PARAMETERS
           Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings): _
         Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings): _
                               Coverage - Caller Response Interval (seconds): _
     Threshold for Blocking Off-Net Redirection of Incoming Trunks Calls: 1

COVERAGE
                                         Keep Held SBA at Coverage Point? _
        External Coverage Treatment for Transferred Incoming Trunk Calls? _
         Immediate Redirection on Receipt of PROGRESS Inband Information? _
                                               Maintain SBA At Principal? _
              QSIG VALU Coverage Overrides QSIG Diversion with Rerouting? _
                                            Station Hunt Before Coverage? n

FORWARDING
                                                   Call Forward Override? _
                                               Coverage After Forwarding? _
```

Screen 237. System-Parameters — Call Coverage / Call Forwarding

Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

This field specifies:

- the number of rings applied at a local coverage point before a call redirects to the next coverage point
- the number of rings applied at the principal before a call forwards when Call Forwarding Busy/Don’t Answer is activated.

Valid entries	Usage
1–99	See note below.

Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

This field specifies:

- the number of rings applied at an off-net coverage point before a call is redirected to the next coverage point
- the number of rings applied at an off-net forwarded-to destination before the call is redirected to coverage.

Valid entries	Usage
1–99	See note below.

NOTE:
When ringing local destinations (say in an office environment), a short interval often is appropriate because the intended party either is near the phone or not present. However, when ringing off-net locations, you cannot assume how near the intended party is to the phone. If the call is left at an off-net destination for only a short interval, the call may be redirected to the next destination before the intended party has any real chance of answering the call.

Coverage - Caller Response Interval (seconds)

The time in seconds the caller (internal caller only) has before a call redirects to the called party's first coverage point. The calling party either can hang up, use Leave Word Calling, or press the GO TO COVER button during this time interval.

Valid entries	Usage
0 through **10**	

Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls

This field applies for those occasions when an incoming call to a station redirects off-net. At that time, the Call Forward timer activates to block any further incoming calls to that station from being redirected off-net until the timer expires.

Valid entries	Usage
1-7	The number of allowed calls to be routed off-net before blocking commences.
all	Call processing never activates the Call Forward timer. Therefore, any number of calls to a principal may be redirected off-net.

Keep Held SBA at Coverage Point

This field governs how a covering user who has placed an answered coverage call on hold is treated if the original principal bridges onto the call.

Valid entries	Usage
y	Keeps the coverage party on the call. The coverage party remains on hold, but may enter the call along with the principal and the calling party.
n	Drops the coverage party from the call.

External Treatment for Transferred Incoming Trunk Calls

This field governs how an transferred incoming trunk call is handled if the call redirects to coverage.

Valid entries	Usage
y	Enter **y** to allow external coverage treatment for incoming trunk calls that redirect to coverage.
n	Enter **n** to allow internal coverage treatment for incoming trunk calls that redirect to coverage

Immediate Redirection on Receipt of PROGRESS Inband Information

This field appears only if one of the following is true:

- The Coverage of Calls Redirected Off-Net Enabled field on the System Parameters Coverage/Forwarding screen is **y**.
- The Value-Added Avaya (VALU) field on the System Parameters Customer Options, Page 6, screen is **y**.

This field pertains only to CCRON and QSIG VALU coverage calls redirected over end-to-end ISDN facilities. Some cellular phone providers send an ISDN PROGRESS message with the Progress Indicator field set to ‘inband information’ when a cellular phone is unavailable to receive a call. In these circumstances, the message indicates that an announcement is being played to the originating party and we should move the call immediately to the next available coverage point.

However, a PROGRESS message with a Progress Indicator of ‘inband information’ may be received for reasons other than an unavailable cellular phone. In this case, you probably do not want to redirect the call to the next coverage point.

There is no way for the Avaya MultiVantage to know the actual intent of such a PROGRESS message, yet you may choose how the system should handle this message. It is essentially an educated, but blind, choice and you should be aware that there will be instances when this choice leads to inappropriate call handling.

Avaya MultiVantage queries this field on receipt of a qualifying PROGRESS message and acts according to your instruction on how to treat it.

As a guide, users in European countries following the ETSI standard and redirecting to GSM cellular phones may want to consider setting this field to **y.**

In the United States, PROGRESS messages with the Progress Indicator field set to 'inband information' are sent for a variety of reasons not associated with unavailable cell phones and you should set this field to **n**.

Valid entries	Usage
y	Immediately redirect an off-net coverage/forwarded call to the next coverage point.
n	Do not immediately redirect an off-net coverage/forwarded call to the next coverage point.

Maintain SBA At Principal

Allows a user to maintain a simulated bridged appearance when a call redirects to coverage.

Valid entries	Usage
y	Enter **y** to maintain a simulated bridged appearance (SBA) on the principal's phone when a call redirects to coverage. DCS with rerouting will not be attempted after coverage.
n	When set to **n**, no SBA is maintained on the principal's phone. DCS with rerouting will be attempted, and if successful, the principal will lose the bridged appearance and the ability to bridge onto the coverage call.

QSIG VALU Coverage Overrides QSIG Diversion with Rerouting

This field specifies whether, with both QSIG Diversion with Rerouting and QSIG VALU turned on, the Coverage After Forwarding option on the Station screen will work for a user for calls that go to remote coverage. Normally, with QSIG Diversion with Rerouting turned on, the local system passes control of a forwarded call to the remote QSIG switch on which the forwarding destination resides. In this case, the forwarded call cannot return to coverage for the user who originally received the call, even when the remote destination is busy or does not answer.

However, you can enter **y** in this field to have QSIG VALU call coverage take precedence. In this case, if the Coverage After Forwarding option on the Station screen is enabled for a user, then QSIG Diversion with Rerouting is not be attempted.

Valid entries	Usage
y/n	Enter **y** to allow Coverage After Forwarding to work when it is activated on a user's Station screen and Diversion with Rerouting is also turned on.

Station Hunt Before Coverage

This field allows you to choose whether a call to a busy station performs station hunting before going to coverage.

Valid entries	Usage
y/n	Enter **y** to use Station Hunt Before Coverage.

Call Forward Override

This field specifies how to treat a call from a forwarded-to party to the forwarded-from party.

Valid entries	Usage
y	Overrides the Call Forwarding feature by allowing a forwarded-to station to complete a call to the forwarded-from station.
n	Directs the system to forward calls to a station even when they are from the forwarded-to party.

Coverage After Forwarding

This field governs whether an unanswered forwarded call is provided coverage treatment.

Valid entries	Usage
y	Coverage treatment is provided to unanswered forwarded calls.
n	No coverage treatment is provided to unanswered forwarded calls. The call remains at the forwarded-to destination.

Field descriptions for page 2

```
change system-parameters coverage-forwarding  (page 2)

          SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)
                            Coverage Of Calls Redirected Off-Net Enabled? y
      Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? y
                                       Ignore Network Answer Supervision? y
                      Disable call classifier for CCRON over ISDN trunks? n
```

Screen 238. System-Parameters Coverage-Forwarding

Coverage Of Calls Redirected Off-Net Enabled

This field allows you to enable/disable the Coverage of Calls Redirected Off-Net (CCRON) feature. This field provides a quick means of disabling this feature if the demand on the call classifier port resources degrades other services provided by the switch. The Coverage of Calls Redirected Off-Net field on this screen must be **y** to administer this field.

Valid entries	Usage
y	Avaya MultiVantage monitors off-net coverage/forwarded calls and provides further coverage treatment for unanswered calls.
n	Avaya MultiVantage does not monitor offnet coverage/forwarded calls. No further coverage treatment is provided if such calls are unanswered.

Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point

This field appears only if the Coverage of Calls Redirected Off-Net Enabled field on this screen is **y**.

When the system redirects a call off-net at the final coverage point in a coverage path, the system can apply no further coverage treatment even if the call is unanswered. The only reason for activating answer detection on such a call is to maintain the simulated bridged appearance (SBA) on the principal's phone that allows the principal to answer or bridge onto the call. However, when the system monitors the call through use of a call classifier port, there is an inherent cut-through delay following the detection of answer at the far end. This field has no consequence when the off-net call is carried end-to-end by ISDN facilities; the SBA is maintained and there is no cut-through delay.

Valid entries	Usage
y	Directs the system to maintain a simulated bridged appearance on the principal when redirecting to a final off-net coverage point.
n	Allows the system to drop the SBA on the principal's phone when the call redirects off-net at the last coverage point, eliminating the cut-through delay inherent in CCRON calls, but sacrificing the principal's ability to answer the call.

Ignore Network Answer Supervision

This field appears only if the Coverage of Calls Redirected Off-Net Enabled field on this screen is **y**.

CCRON may use a call classifier port to determine whether an off-net coverage or forwarded call has been answered, discarding other information that may indicate an answered state. However, some customers pay the operating company to provide network answer supervision on their trunks and desire that CCRON not discard that information. They may preserve this service by setting this field to **n**.

On the other hand, beware when you tandem a call over a tie trunk through another switch node from where it redirects to the public network over non-ISDN facilities. If the trunk on the far-end node sends a timed answer supervision, that may get tandemed back to the originating switch as a network answer. In such a scenario, the originating switch interprets the call as answered, leading to some undesirable behavior. To avoid these calls from mistakenly be treated as answered, set this field to **y**. An unfortunate consequence is that a short cut-through delay that is inherent to call classification is introduced when the call is answered.

Valid entries	Usage
y	Ignore network answer supervision and rely on the call classifier to determine when a call is answered.
n	Treat network answer supervision as a true answer.

Disable call classifier for CCRON over ISDN trunks

When a CCRON call routes offnet over ISDN end-to-end facilities, no call classifier is attached to the call. If, subsequently during the call, an ISDN PROGRESS or ALERT message is received that indicates that interworking has occurred, a call classifier is normally attached to the call and assumes precedences over ISDN trunk signalling. This field provides a customer the means of directing the switch to dispense with the call classifier on interworked calls and rely on the ISDN trunk signalling messages.

Valid entries	Usage
y	Use **y** to disable the call classifier for CCRON calls over interworked trunk facilities.
n	Use **n** to enable the call classifier for CCRON calls over interworked trunk facilities.

System Parameters Country-Options

This screen implements parameters associated with certain international (including North American) call characteristics. You cannot change this screen. See your Avaya representative if you want to modify any of the values here.

Refer to the following table for country codes throughout this screen.

Country code table

Code	Country
1	United States, Canada
2	Australia
3	Japan
4	Italy
5	Netherlands
6	Singapore
7	Mexico
8	Belgium, Luxembourg
9	Saudi Arabia
10	United Kingdom

Code	Country
11	Spain
12	France
13	Germany
14	Czech Republic, Slovakia
15	Russia (CIS)
16	Argentina
17	Greece
18	China
19	Hong Kong
20	Thailand
21	Macedonia
22	Poland
23	Brazil
24	Nordic
25	South Africa

Field descriptions for page 1

```
change system-parameters country-options                          Page 1 of 7

                        SYSTEM PARAMETERS COUNTRY-OPTIONS


              Companding Mode: Mu-Law         Base Tone Generation Set: 1
          440Hz PBX-dial Tone? n            440Hz Secondary-dial Tone? n
       Analog Ringing Cadence: 1    Set Layer 1 timer T1 to 30 seconds? n
     Analog Line Transmission: 1
        Display Character Set? Roman
       Howler Tone After Busy: y  Disconnect on No Answer by Call Type: y
              Enable Busy Tone Disconnect for Analog Loop-start Trunks?

TONE DETECTION PARAMETERS
        Tone Detection Mode: 5          Dial Tone Validation Timer:
             Interdigit Pause: short
```

Screen 239. System Parameters Country-Options

Companding Mode

Identifies the companding algorithm to be used by system hardware.

Valid entries	Usage
A-law	Generally used outside the US
Mu-law	Generally used in the US

Base Tone Generation Set

The country code identifies the base tone generation set to be used. This field is meaningful only if the system tone detector is a TN780, vintage 4 or a TN2182.

A TN780 vintage 5 or greater or a TN2182 is required if Belgian Tones (Country code 8) are specified.

Valid entries	Usage
1–**25**	Refer to the Country code table at the beginning of this screen description.

440Hz PBX-dial Tone

Specifies whether the switch (primary) dial tone will be changed to a continuous 440Hz/-17 tone.

Valid entries	Usage
y/n	A value of **n** implies the tone will either be administered on a later page of this screen or, if no individual definition is administered, as defined in Base Tone Generation Set.

440Hz Secondary-dial Tone

Specifies whether the Secondary (CO) dial tone will be changed to a continuous 440Hz/-17 tone.

Valid entries	Usage
y/n	A value of **n** implies the tone will either be administered on a later page of this screen or, if no individual definition is administered, as defined in Base Tone Generation Set.

Analog Ringing Cadence

The country code identifies the ringing cadence to be used by analog phones in the system.

Valid entries	Usage
1 to **25**	Refer to the Country code table at the beginning of this screen description. For more information on, see Audible Ringing Patterns in *Avaya MultiVantage™ Hardware Solutions Guide*.

NOTE:
If the Analog Ringing Cadence is set to 3, 10, 11, or 25, then the Message Waiting Indicator on K2500 phones does not flash.

Set Layer 1 timer T1 to 30 seconds

Valid entries	Usage
y/n	Specifies whether the Layer 1 timer is set to 30 seconds.

Analog Line Transmission

The country code identifies the transmission and signaling parameters.

Valid entries	Usage
1–**25**	Refer to the Country code table at the beginning of this screen description.

Display Character Set

Valid entries	Usage
cyrillic **katakana** **roman** **ukranian**	Indicate the enhanced character set to display. See Enhanced Telephone Display for more information. **NOTE:** Cyrillic, Roman, and Ukranian map to the Eurofont character set.

Howler After Busy

Plays howler tone when users leave their analog phone off-hook too long.

Valid entries	Usage
y/n	Enables howler tone.

Disconnect on No Answer by Call Type

Drops outgoing trunk calls (except DCS and AAR) that users leave unanswered too long.

Valid entries	Usage
y/n	Enables the system to disconnect calls that are not answered.

Enable Busy Tone Disconnect for Analog Loop-start Trunks

This field allows the switch to recognize a busy tone from the central office as a disconnect signal.

Valid entries	Usage
y/n	Enter **y** to allow the switch to disconnect the trunk when a busy tone is received from the central office.

Tone Detection Mode

The country code specifies the type of tone detection used on a TN420B (or later) tone-detection circuit pack.

Valid entries	Usage
1	Precise Italian tone detection algorithm
2	Precise Australian tone detection algorithm
3	Precise UK tone detection algorithm
4	Imprecise normal broadband filter algorithm (valid with TN420C or later Tone Detector circuit pack)
5	Imprecise wideband filter algorithm (valid with TN420C or later Tone Detector circuit pack)
6	

Dial Tone Validation Timer

Displays number of milliseconds that the dial tone validation routine will use to sample transmissions. This field appears only when Tone Detection Mode is equal to **4** or **5**. (Valid with TN420C or later Tone Detector circuit pack.)

Valid entries	Usage
0–**6375** in increments of 25	
600	

Interdigit Pause

Specifies the maximum length of the inter-digit pause. Breaks lasting less than this range will be bridged or ignored. (Valid with TN420C or later Tone Detector circuit pack.)

Valid entries	Usage
short	5 to 30ms
long	20 to 40ms

Field descriptions for page 2

```
display system-parameters country-otpions                         Page 2 of 7
                                 2 PARTY LOSS PLAN

        Digital Loss Plan: 1                                 Customize? n
                                           TO
          1   2   3   4   5   6   7   8   9  10  11  12  13  14  15  16  17
     1:  -1   0   0   0   3   0   0   0   0   3   0   6   6   6   0   3   3
     2:   0   0   0   0   0   3   3   2   2   3   0   6   6   6   2   3   3
     3:   0   0   0   0   0   3   3   3   2   3   0   6   6   6   0   3   3
     4:  15   0   0   0   6   0   0   0   0   3   0   6   6   6   0   3   3
     5:   0  -3  -3   0   0  -3  -3  -3  -3   0  -3   3   0   0  -3   3   3
     6:   0   3   3   0   0   6   8   6   5   5   5   9   9   9   5   3   3
F    7:   0   3   3   0   0   8   8   6   5   5   5   9   9   9   5   3   3
R    8:   0   3   3   0   0   6   6   6   3   5   3   9   6   6   3   3   3
O    9:   0   2   2   0   0   5   5   3   0   0   2   3   3   3   9   3   3
M   10:   3   3   3   3   3   5   5   5   0   0   3   3   3   3   3   3   3
    11:   0   0   0   0   0   5   5   3   2   3   0   6   6   3   0   3   3
    12:   0   0   0   0   0   3   3   3  -3  -3   0   0   0   0   0   3   3
    13:   0   0   0   0   0   3   3   3  -3  -3   0   0   0   0   0   3   3
    14:   0   0   0   0   0   3   3   3  -3  -3  -3   0   0   0   0   3   3
    15:   0   2   0   0   0   5   5   3   0   3   0   6   6   6   0   3   3
    16:   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3
    17:   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3   3
```

Screen 240. System Parameters Country-Options

Digital Loss Plan

Provides the default values for digital loss plan and n-party conference loss.

Valid entries	Usage
1 - **25**	Refer to the Country code table at the beginning of this screen description.

Customize

Appears when the Digital Loss Plan Modification field is enabled on the System-Parameters Customer-Options screen.

Valid entries	Usage
y/n	Enables customization on the 2-party loss table.

FROM / TO

Identifies the variable digital loss values.

Valid entries	Usage
-3 - 15	An unsigned number is a loss, while a number preceded with a minus sign is a gain.

Field descriptions for page 3

```
display system-parameters country-options                        Page 3 of 7
                              TONE & CONFERENCE LOSS PLANS

        Digital Tone Plan: 1                          Customize? n
                                     TO
              1   2   3   4   5   6   7   8   9  10  11  12  13  14  15  16  17
      Dial:   0   3   3   0   0   6   6   6   5   0   6   5   5   5   5   0   0
   Confirm:   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
   Reorder:   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
      Busy:   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
   Ringing:   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
 Spec Ring:   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
 Intercept:   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
   Waiting:   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
    Verify:   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
   Intrude:   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0   0
       Zip:   3   3   3  -3  -3  -3  -3  -3  -3  -3  -3  -3  -3  -3  -3   0   0
     Music:   0   3   3   0   0   6   6   6   3   0   6   3   3   3   3   0   0


End-to-End total loss (dB) in a n-party conference:
3:____      4:____      5:____      6:____
```

Screen 241. System Parameters Country-Options

Digital Tone Plan

Provides the default values for tone loss for the selected country.

Valid entries	Usage
1 - **25**	Refer to the Country code table at the beginning of this screen description.

Customize

Appears when, on the System-Parameters Customer-Options screen, the Digital Loss Plan Modification field is **y.**

Valid entries	Usage
y/n	Enables customization on the 2-party loss table.

End-to-End total loss (dB) in a n-party conference

Provides total loss for a conference call with the designated number of parties.

Valid entries	Usage
0 - 99	The higher the number listed for a call with a fixed display number of parties, the more loss the switch adds into a conference call with that number of parties; therefore, the conference call is quieter.

FROM / TO

Identifies the variable digital tone values.

Valid entries	Usage
-3 - 15	An unsigned number is a loss, while a number preceded with a minus sign is a gain.

Field descriptions for pages 4–7

```
change system-parameters country-options                       Page 4 of 7
                      SYSTEM PARAMETERS COUNTRY-OPTIONS
         Tone Name     Cadence       Tone
                        Step    (Frequency/Level)
      busy____________    1:      440/-17.25___   Duration (msec): 200__
                          2:      silence______   Duration (msec): 200__
                          3:      440/-17.25___   Duration (msec): 200__
                          4:      silence______   Duration (msec): 200__
                          5:      440/-17.25___   Duration (msec): 200__
                          6:      goto_________              step: 3_
                          7:      _____________
                          8:      _____________
                          9:      _____________
                         10:      _____________
                         11:      _____________
                         12:      _____________
                         13:      _____________
                         14:      _____________
                         15:      _____________
```

Screen 242. System Parameters Country-Options

Tone Name

Each entry uses one of the keywords below to indicate which of the individually administrable tones this screen modifies. This field (with its associated Tone (Frequency/Level), Duration, and Step fields) is meaningful only if the system tone detector is a TN780, vintage 4 or greater, or a TN2182.

Valid entries	Usage
blank	If this field is blank, all entries in the corresponding Frequency and Duration fields are ignored.
1-call-wait **2-call-wait** **3-call-wait** **busy** **busy-verify** **call-wait-ringback** **conference** **confirmation** **disable-dial** **hold** **hold-recall** **immed-ringback** **intercept** **intrusion** **PBX-dial** **recall-dial** **recall-dont-ans** **redirect** **reorder** **rep-confirmation** **reset-shift** **ringback** **secondary-dial** **special-dial** **whisper-page** **zip**	

Cadence Step

Valid entries	Usage
1–15	Identifies the number of each tone cadence step.

Tone (Frequency/Level)

Valid entries	Usage
silence	An entry of **silence** means no tone. A final step of **silence** with an infinite duration will be added internally to any tone sequence that does not end in a **goto**.
goto	An entry of **goto** means to repeat all or part of the sequence, beginning at the specified cadence step.
350/-17.25 350+425/-4.0 350+440/-13.75 375+425/-15.0 404/-11.0 404/-16.0 404+425/-11.0 404+450/-11.0 425/-4.0 425/-11.0 425/-17.25 440/-17.25 440+480/-19.0 450/-10 480/-17.25 480+620/-24.0 525/-11.0 620/-17.25 697/-8.5 770/-8.5 852/-8.5 941/-8.5 1000/0.0 1000/+3.0 1004/0.0 1004/-16.0 1209/-7.5 1336/-7.5 1400/-11.0 1477/-7.5 1633/-7.5 2025/-12.1 2100/-12.1 2225/-12.1 2804/-16.0	Specifies the frequency and level of the tone.

Duration (msec)

There is one dynamic Duration field associated with each of the 15 Tone (Frequency/Level) fields on each screen page. Initially, when Tone is blank, this field does not appear. However, when a non-blank value other than **goto** is entered in a Tone field, the associated Duration field appears, and must be used to specify the duration (in milliseconds) of the specified tone.

Valid entries	Usage
50 through **12750** in increments of 50	To describe the duration of each administered tone.

Step

There is one dynamic Step field associated with each of the 15 Tone (Frequency/Level) fields shown on a screen page. Initially, when Tone is blank, this field does not appear. However, when **goto** is entered in a Tone field, the associated Step field appears, and must be used to specify the cadence step to begin repeating from

Valid entries	Usage
Cadence Step (**1**–**14**)	Beginning the repeated sequence for a “**goto**” entry.

System-Parameters Customer-Options

This screen shows you which optional features are enabled for your system as determined by the installed license file. If you have any questions about disabling or enabling one of these features contact your Avaya representative.

Field descriptions for page 1

```
display system-parameters customer-options  (page 1)
                              OPTIONAL FEATURES
                                                                 Used
        G3 Version: V10                    Maximum Ports: 2800   1041
          Location: 1           Maximum XMOBILE Stations: 0      0
          Platform:

IP PORT CAPACITIES
                          Maximum Administered IP Trunks: 100    96
             Maximum Concurrently Registered IP Stations: 10     10
               Maximum Administered Remote Office Trunks: 0      0
  Maximum Concurrently Registered Remote Office Stations: 0      0
                Maximum Concurrently Registered IP eCons: 0      0

     Maximum Number of DS1 Boards with Echo Cancellation: 0      0
                                      Maximum VAL Boards: 1      0
           (NOTE: You must logoff & login to effect the permission changes.)
```

Screen 243. System Parameters Customer-Options

G3 Version

Identifies the version of Avaya MultiVantage being used.

Maximum Ports

Number of ports active, per contract.

Location

Indicates the location of this switch. 1 indicates Canada or the United States. 2 indicates any other location, and allows the use of International Consolidation circuit packs and telephones.

Maximum XMOBILE Stations

Specifies the maximum number of allowable XMOBILE stations. In general, each XMOBILE station is assigned to a wireless handset. Each XMOBILE station counts as a station and a port in terms of system configuration.

Platform

A display-only field that identifies, via a number mapping, the platform being used for a specific customer.

Maximum Administered IP Trunks

Defines limits of the number of IP trunks administered.

Maximum Concurrently Registered IP Stations

Specifies the maximum number of IP stations that may be registered at one time. Must be less than or equal to the Maximum Ports field on this page. The switch will use the smaller of this number or the number based on product ID on page 7 of this screen.

Maximum Administered Remote Office Trunks

Defines limits of the number of IP endpoints based on the endpoint. Use the smaller of this number or the number based on product ID on page 7 of this screen.

Maximum Concurrently Registered Remote Office Stations

Specifies the maximum number of remote office stations that may be registered at one time. Must be less than or equal to the Maximum Ports field on this page. The switch will use the smaller of this number or the number based on product ID on page 7 of this screen.

Maximum Concurrently IP eCons

Specifies the maximum number of IP SoftConsoles that may be registered at one time. The maximum number depends on the type of system.

Maximum Number of DS1 Boards with Echo Cancellation

Shows the number of DS1 circuit packs that can have echo cancellation.

Maximum VAL Boards

Valid entries	Usage
0–10 (DEFINITY Server R only) **0-5** (all others)	This display-only field indicates the maximum number of TN2501AP (Voice Announcement over LAN) boards allowed in this system. ■ For values greater than 1, the Val Full 1-Hour Capacity field on page 4 of the System-Parameters Customer-Options screen must be set to **y**. ■ This field updates the System Limit field on the System Capacity report.

Field descriptions for page 2

```
display system-parameters customer-options  (page 2)
                      OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List?            Audible Message Waiting?
        Access Security Gateway (ASG)?               Authorization Codes?
        Analog Trunk Incoming Call ID?                        CAS Branch?
 A/D Grp/Sys List Dialing Start at 01?                          CAS Main?
Answer Supervision by Call Classifier?                 Change COR by FAC?
                                  ARS?    Computer Telephony Adjunct Links?
                 ARS/AAR Partitioning?         Co-Res DEFINITY LAN Gateway?
          ARS/AAR Dialing without FAC?     Cvg Of Calls Redirected Off-net?
          ASAI Link Core Capabilities?                         DCS (Basic)?
          ASAI Link Plus Capabilities?                   DCS Call Coverage?
       Async. Transfer Mode (ATM) PNC?                  DCS with Rerouting?
  Async. Transfer Mode (ATM) Trunking?
                                            Digital Loss Plan Modification?
                                 ATMS?                             DS1 MSP?
                  Attendant Vectoring?                DS1 Echo Cancellation?
```

Screen 244. System Parameters Customer-Options

Abbreviated Dialing Enhanced List

Provides the capability to store and retrieve dialing lists that simplify or eliminate dialing. You dial an abbreviated code or depress an assigned button. The stored entries are organized in number lists. There are three types of number lists: personal, group, and enhanced.

Access Security Gateway (ASG)

Provides an additional level of security for remote administration of the switch.

A/D Grp/Sys List Dialing Start at 01

Allows you to number Abbreviated Dialing group or system lists starting with 01, rather than simply 1. This allows Avaya MultiVantage Abbreviated Dialing to operate like it does with the DEFINITY G2 system.

Analog Trunk Incoming Call ID

This field allows collection and display the name and number of an incoming call information on analog trunks.

Answer Supervision by Call Classifier

This circuit pack detects tones and voice-frequency signals on the line and determines whether a call has been answered. This field is set to **y** if the system contains a call-classifier circuit pack.

ARS

Provides access to public and private communications networks. Long-distance calls can be routed over the best available and most economical routes. Provides partitioning of ARS routing patterns.

ARS/AAR Partitioning

Provides the ability to partition AAR and ARS into 8 user groups within a single Avaya MultiVantage. Can establish individual routing treatment for each group.

ARS/AAR Dialing without FAC

Provides for Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) calls without dialing a Feature Access Code (FAC).

ASAI Link Core Capabilities

Provides linkage between Avaya MultiVantage and adjuncts. CallVisor ASAI improves the call handling efficiency of ACD agents and other system users by allowing an adjunct to monitor, initiate, control, and terminate calls on the switch.

NOTE:
ASAI Link Core Capabilities only applies to links administered as type asai. This field was previously named ASAI Interface.

If the ASAI Link Core Capabilities field is administered to **y** then it will be associated with the following ASAI capability groups:

- Adjunct Control
- Domain Control
- Event Notification
- Single Step Conference
- Request Feature
- II Digits
- Set Value
- Value Query

For more information, see the *Avaya MultiVantage CallVisor ASAI Technical Reference*.

ASAI Link Plus Capabilities

Provides linkage between Avaya MultiVantage and adjuncts. If the ASAI Link Plus Capabilities field is administered to **y** then it will be associated with the following ASAI capability groups:

- Adjunct Routing
- Answering Machine Detection
- Selective Listening
- Switch Classified Outbound Calls

NOTE:
ASAI Link Plus Capabilities only applies to links administered as type asai.

For more information, see the *Avaya MultiVantage CallVisor ASAI Technical Reference*.

Asynch. Transfer Mode (ATM) PNC

ATM PNC can be enabled only if:

- all prior fiber-link administration has been removed
- all “switch-node” and “dup-switch-node” carrier types have been removed

Asynch. Transfer Mode (ATM) Trunking

If ATM trunking is enabled, multiple ISDN-PRI T1 or E1 trunks can be emulated on one ATM pipe. Can only be enabled if the ISDN-PRI field is set to **y**. Enables circuit emulation service (CES).

ATM WAN Spare Processor

An ATM WAN spare processor acts as a PPN in the event of network failure, and can function as an SPE if the main PPN is not functional.

ATMS

Provides for voice and data trunk facilities to be measured for satisfactory transmission performance.

Attendant Vectoring

Allows you to use attendant vectoring. Cannot be set to **y** if the CAS Main field is **y**.

Audible Message Waiting

Provides audible message waiting.

Authorization Codes

Permits you to selectively specify levels of calling privileges that override in-place restrictions. In addition to facilities access, authorization codes are used for unique identification for billing security purposes.

CAS Branch

Provides Centralized Attendant Service - Branch. See CAS Main for more information.

CAS Main

Provides multi-location switch customers served by separate switching vehicles to concentrate attendant positions at a single main Avaya MultiVantage location. The main Avaya MultiVantage is served by an attendant queue that collects calls from all locations (main and branch). Each branch location switches all of its incoming calls to the centralized attendant positions over release link trunks (RLTs). The calls are then extended back to the requested extension at the branch switch over the same RLT. When the call is answered, the trunks to the main switch are dropped and can be used for another call.

Change COR by FAC

Provides certain users the ability to change the class of restriction of local extensions and local attendants via a phone by using a feature access code (FAC). Cannot be set to **y** if the Tenant Partitioning field is **y**.

Co-Res DEFINITY LAN Gateway

(For S8100 Media Server and S8300 Media Server only.) Provides communications between TCP/IP clients and MultiVantage call processing.

For more information see the *Avaya MultiVantage CallVisor ASAI Technical Reference*.

Computer Telephony Adjunct Links

Provides linkage between Avaya MultiVantage and adjuncts. Includes both the ASAI Link Core and ASAI Link Plus capabilities, plus the Phantom Calls and CTI Stations.

NOTE:
Computer Telephony Adjunct Links only applies to links administered as type adjlk. This field was previously named ASAI Proprietary Adjunct Links.

For more information see the *Avaya MultiVantage CallVisor ASAI Technical Reference*.

Cvg Of Calls Redirected Off-net

Provides continued monitoring for calls redirected to off-network (remote) coverage points. Uses call classification via call classifier circuit pack or ISDN trunk signaling.

DCS (Basic)

Provides transparent operation of selected features across a Distributed Communications System (DCS). Users on one switch can use features located on another switch. Includes 4- and 5-digit uniform dialing and 1–4 digit steering. Does not support a 6/7-digit dial plan.

DCS Call Coverage

Provides DCS-based transparency of the call coverage feature across a DCS network of switches.

DCS with Rerouting

Provides for rerouting calls transferred among DCS nodes, enabling rerouting of the call for more effective use of facilities.

Digital Loss Plan Modification

Allows you to customize the digital loss and digital tone plans.

DS1 MSP

Provides the ability to change fields on DS1 circuit pack screen without removing the related translations of all trunks from the trunk group.

DS1 Echo Cancellation

Removes perceivable echo from the system.

Field descriptions for page 3

```
               display system-parameters customer-options  (page 3)

                                   OPTIONAL FEATURES

  Emergency Access to Attendant?                          ISDN-BRI Trunks?
         Enhanced Conferencing?                                  ISDN-PRI?
                Enhanced EC500?                      Local Spare Processor?
        Extended Cvg/Fwd Admin?                       Malicious Call Trace?
   External Device Alarm Admin?    Mode Code for Centralized Vocie Mail?
              Flexible Billing?
  Forced Entry of Account Codes?                  Multifrequency Signaling?
     Global Call Classification?  Multimedia Appl.Server Interface(MASI)?
           Hospitality (Basic)?       Multimedia Call Handling (Basic)?
Hospitality (G3V3 Enhancements)?     Multimedia Call Handling(Enhanced)?
                     IP Trunks?                        Multiple Locations?
         IP Attendant Consoles?         Personal Station Access (PSA?)?
                   IP Stations?

             ISDN Feature Plus?
  ISDN Network Call Redirection?
```

Screen 245. System Parameters Customer-Options

Emergency Access to Attendant

Provides for emergency calls to be placed to an attendant. These calls can be placed automatically by Avaya MultiVantage or dialed by users.

Enhanced Conferencing

Enhanced Conferencing allows the customer to use the Meet-me Conference, Selective Conference Party Display, Drop, and Mute, and No Hold Conference features. Must be **y** for Enhanced Conferencing.

Enhanced EC500

Provides mobile call services including "Anytime Anywhere" accessibility with One Number availability and Origination mapping.

Extended Cvg/Fwd Admin

Provides basic telecommuting package capability for Extended User Administration of Redirected Calls.

External Device Alarm Admin

Provides for analog line ports to be used for external alarm interfaces. Allows identification of port location, adjunct associated with port location, and the alarm level to report.

Flexible Billing

Provides an internationally accepted standard interface for end-to-end digital connectivity. Used with a T1 interface and supports twenty-three 64-KBPS voice or data B-Channels and one 64-Kbps signaling D Channel for total bandwidth of 1.544 Mbps.

Forced Entry of Account Codes

Allows system administration to force account users to enter account codes based on user or trunk class of restriction, or by an option on the Toll Analysis table. FEAC provides an easy method of allocating the costs of specific calls to the correct project, department, etc.

Global Call Classification

Provides call classification outside of North America. Listens for tones and classifies tones detected. Required for Call Coverage Off Net and Outgoing Call Management.

Hospitality (Basic)

Provides access to basic features including: Attendant Crisis Alert, Attendant Room Status, Automatic Wakeup, Custom Selection of VIP DID Numbers, Do Not Disturb, Names Registration, Single-Digit Dialing, and Mixed Station Numbering.

Hospitality (G3V3 Enhancements)

Software required for Property Management System and Automatic Wakeup. Property Management System Interface activates Forward PMS Messages to INTUITY Lodging and PMS Protocol Mode (transmit in ASCII mode).

Usage Note: standard hospitality features are included in basic system software.

Internet Protocol (IP) PNC

This field is only for the Avaya S8700 Media Server for IP Connect Configurations and then should be set to **y**. This field cannot be administered if:

- the Asynch. Transfer Mode (ATM) PNC field is **y**
- the PNC Duplication field is **y**
- If fiber is administered

IP Trunks

Controls permission to administer H.323 trunks. Must be **y** for IP trunks.

IP Attendant Consoles

Controls permission to administer the IP Attendant Console.

IP Stations

Controls permission to administer H.323 and/or SoftPhone stations. Must be **y** for IP phones.

ISDN Feature Plus

Provides ISDN Feature Plus signaling. This option is enabled when either the ISDN-BRI Trunks field or the ISDN-PRI field is **y**.

ISDN Network Call Redirection

Administrable if the ISDN-PRI or ISDN-BRI Trunk field is **y**. Network Call Redirection (NCR) redirects an incoming ISDN call from an Avaya MultiVantage system to another PSTN endpoint. Used in Call Centers with Best Service Routing and Lookahead Interflow. For more information, see *Avaya MultiVantage Network Call Redirection* (555-233-759).

ISDN-BRI Trunks

Provides the capability to add ISDN-BRI trunks to the switch. If enabled, can add isdn trunk groups and the following screens are accessible:

- network-facilities
- private-numbering
- public-unknown- numbering

ISDN-PRI

Provides Integrated Services Digital Network (ISDN-PRI) software for either a switch-hardware platform migration only or a switch-hardware platform migration in combination with a software version upgrade. Also provides signaling support for H.323 signaling. Must be **y** for IP trunks.

Local Spare Processor

A display-only field that indicates that the switch is a Local Spare Processor (LSP).

Malicious Call Trace

Provides ability to retrieve certain information related to a malicious call.

Mode Code for Centralized Voice Mail

This feature provides the ability to share a Voice Mail System among several switches using the Mode Code - Voice Mail System Interface.

Multifrequency Signaling

Provides for a screen of number signaling used between the switch and the central office.

Multimedia Appl. Server Interface (MASI)

Allows users of the Multimedia Communications Exchange (MMCX) to take advantage of certain Avaya MultiVantage telephony features.

Multimedia Call Handling (Basic)

Allows administration of desktop video-conferencing systems as data modules associated with Avaya MultiVantage voice stations in a multimedia complex. Users can dial one number to reach either endpoint (voice or data) in the complex. Also provides support for IP SoftPhones.

Multimedia Call Handling (Enhanced)

Allows a multifunction phone to control a multimedia call like a standard voice call.

Multiple Locations

Allows you to establish numbering plans and time zone and daylight savings plans that are specific for each cabinet in a port network.

Personal Station Access (PSA)

Provides basic telecommuting package capability for Personal Station Access.

Field descriptions for page 4

```
display system-parameters customer-options (page 4)
                                OPTIONAL FEATURES

                  PNC Duplication? n                   Tenant Partitioning? n
             Port Network Support? y           Terminal Trans. Init. (TTI)? y
                                                       Time of Day Routing? y
         Processor and System MSP? n                  Uniform Dialing Plan? y
               Private Networking? y         Usage Allocation Enhancements? y
                                                  VAL Full 1-Hour Capacity? y

                    Remote Office? n                    Wideband Switching? y
    Restrict Call Forward Off Net? y                              Wireless? n
            Secondary Data Module? y
            Station and Trunk MSP? n
     Station as Virtual Extension? n
      Survivable Remote Processor? n
  System Management Data Transfer? n
```

Screen 246. System Parameters Customer-Options

PNC Duplication

This field only appears on Avaya DEFINITY Server R. If set to **y**, the Enable Operation of PNC *(Port Network Connectivity)* Duplication field appears on the System Parameters Duplication screen. The Enable Operation of PNC Duplication field is set with the Enable Operation of SPE *(Switch Processing Element)* Duplication field to provide non-standard reliability levels (high, critical, or ATM PNC Network Duplication).

Port Network Support

Indicates that the switch is operating as a stand-alone Internal Communications Controller (ICC) when set to **n** and is used to disable traditional port networking. Set to **y** to indicate that traditional DEFINITY port networks are in use.

Processor and System MSP

Allows the customer administrator or technician to maintain processor and system circuit packs.

Private Networking

Upgrades PNA or ETN software RTU purchased with earlier systems.

VAL Full 1-Hour Capacity

If this is enabled, you have the Enhanced offer, which allows up to 60 minutes storage capacity per pack and multiple integrated announcement circuit packs.

Remote Office

Allows administration of a remote office.

Restrict Call Forward Off Net

The system can monitor the disposition of an off-call and, if it detects busy, bring the call back for further processing, including call coverage.

Secondary Data Module

Provides ability to use any data module as a secondary data module.

Station and Trunk MSP

Provides the customer administrator or technician to maintain station and trunk circuit packs.

Station as Virtual Extension

Allows **virtual** to be entered in the type field of the station screen, which allows multiple virtual extensions to be mapped to a single physical analog phone. The user can also administer a specific ringing pattern for each virtual extension. Useful in environments such as college dormitories, where three occupants can have three different extensions for one physical phone.

Survivable Remote Processor

This field can be set to **y** only if, on the System-Parameters Maintenance screen, the CPE Alarm Activation Level field is **none**. This applies only to DEFINITY Server SI systems.

System Management Data Transfer

Indicates the switch is accessible by Network Administration.

Tenant Partitioning

Provides for partitioning of attendant groups and/or stations and trunk groups. Typically this is used for multiple tenants in a building or multiple departments within a company or organization.

Terminal Trans. Init. (TTI)

Allows administrators of Terminal Translation Initialization (TTI) to merge an station administered with X in the Port field, to a valid port by dialing a system-wide TTI security code and the extension from a terminal connected to that port. Must be set to **y** for Automatic Customer Telephone Rearrangement.

Time of Day Routing

Provides AAR and ARS routing of calls based on the time of day and day of the week. You can take advantage of lower calling rates during specific times.

Uniform Dialing Plan

Provides 3- to 7–digit Uniform Dial Plan (UDP) and 1–7 digit steering. Also allows you to use Extended Trunk Access and Extension Number Portability features.

Usage Allocation Enhancements

Provides for assigning ISDN-PRI or ISDN-BRI Services/Features for Usage Allocation Plans. To use this enhancement, first set either the ISDN-PRI or ISDN-BRI Trunks fields to **y**.

Wideband Switching

Provides wideband data software for switching video or high-speed data. You can aggregate DSO channels up to the capacity of the span. Wideband supports H0, H11, and H12 standards, where applicable, as well as customer-defined data rates.

Wireless

Provides right to use for wireless applications in certain Network Systems sales. You may purchase it from Avaya Network Wireless Systems.

Field descriptions for Call Center Optional Features

```
display system-parameters customer-options (page 5)

                     CALL CENTER OPTIONAL FEATURES

                       Call Center Release:

                             ACD? y     PASTE (Display PBX Data on Phone)? n
                    BCMS (Basic)? y                          Reason Codes? n

      BCMS/VuStats Service Level? n            Service Observing (Basic)? y
               Business Advocate? n     Service Observing (Remote/By FAC)? n
                 Call Work Codes? y             Service Observing (VDNs)?
     DTMF Feedback Signals For VRU? y                           Timed ACW?
                Dynamic Advocate? n                    Vectoring (Basic)? y
    Expert Agent Selection (EAS)? y                Vectoring (Prompting)? y
                        EAS-PHD? n            Vectoring (G3V4 Enhanced)?
               Forced ACD Calls? n     Vectoring (ANI/II-Digits Routing)? n
            Least Occupied Agent?      Vectoring (G3V4 Advanced Routing)? n
       Lookahead Interflow (LAI)?                      Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n  Vectoring (Best Service Routing)? n
    Multiple Call Handling (Forced)?n                Vectoring (Holidays)?
```

Screen 247. Call Center Optional Features

ACD

Automatic Call Distribution (ACD) automatically distributes incoming calls to specified splits or skills. Provides the software required for the Call Center Basic, Plus, Deluxe, and Elite features for the number of agents specified.

BCMS (Basic)

Provides real-time and historical reports about agent, ACD split, Vector Directory Number (VDN) and trunk group activity.

BCMS/VuStats Service Level

Allows you to set up hunt groups or Vector Directory Numbers (VDNs) with an acceptable service level. An acceptable service level defines the number of seconds within which a call must be answered to be considered acceptable.

Business Advocate

Software that provides an integrated set of advanced features to optimize call center performance. For information on CentreVu® Advocate, contact your Account Executive.

Call Center Release

Displays the call center release installed on the system.

Call Work Codes

Allows agents to enter digits for an ACD call to record customer-defined events such as account codes or social security numbers.

DTMF Feedback Signals For VRU

Provides support for the use of C and D Tones to VRUs.

Dynamic Advocate

Software that provides an integrated set of advanced features to optimize call center performance.

EAS-PHD

Increases the number of skills an agent can log in to from four to 20. Increases the number of agent skill preference levels from two to 16.

Expert Agent Selection (EAS)

Provides skills-based routing of calls to the best-qualified agent.

Forced ACD Calls

See Multiple Call Handling.

Least Occupied Agent

Allows call center calls to be routed to the agent who has been the least busy, regardless of when the agent last answered a call.

Lookahead Interflow (LAI)

Provides Look-Ahead Interflow to balance the load of ACD calls across multiple locations.

Multiple Call Handling (On Request)

Allows agents to request additional calls when active on a call.

Multiple Call Handling (Forced)

Forces an agent to be interrupted with an additional ACD call while active on an ACD call. Splits or skills can be one forced, one per skill, or many forced.

PASTE (Display PBX Data on Phone)

Provides an interface between the display of a DCP telephone set and PC-based applications.

Reason Codes

Allows agents to enter a numeric code that describes their reason for entering the AUX work state or for logging out of the system.

Service Observing (Basic)

Allows a specified user to observe an in-progress call on a listen-only or listen-and-talk basis.

Service Observing (Remote/By FAC)

Allows users to service observe calls from a remote location or a local station using this feature’s access codes.

Service Observing (VDNs)

Provides the option of observing and/or monitoring another user’s calls.

Timed ACW

Places an auto-in agent in ACW for an administered length of time after completion of the currently active ACD call.

Vectoring (Basic)

Provides basic call vectoring capability.

Vectoring (Prompting)

Allows flexible handling of incoming calls based on information collected from the calling party or from an ISDN-PRI message.

Vectoring (G3V4 Enhanced)

Allows the use of enhanced comparators, wildcards in digit strings for matching on collected digits and ANI or II-digits, use of Vector Routing Tables, multiple audio/music sources for use with wait-time command and priority level with the oldest-call-wait conditional.

Vectoring (ANI/II-Digits Routing)

Provides for ANI and II-Digits vector routing.

Vectoring (G3V4 Advanced Routing)

Provides for Rolling Average Speed of Answer Routing, Expected Wait Time Routing, and VDN Calls Routing.

Vectoring (CINFO)

Provides the ability to collect ced and cdpd from the network for vector routing. To use this enhancement, first set either the ISDN-PRI or ISDN-BRI Trunks fields to **y**.

Vectoring (Best Service Routing)

Enables the Best Service Routing™ feature. Through special vector commands, Best Service Routing allows you to compare splits or skills at local and remote locations and queue a call to the resource that will give the caller the best service.

Vectoring (Holidays)

Enables the Holiday Vectoring feature. It simplifies vector writing for holidays.

Field descriptions for page Call Center Optional Features

```
                                                            Page X of X
                         CALL CENTER OPTIONAL FEATURES

        VDN of Origin Announcement? n                             VuStats? n
            VDN Return Destination? n                VuStats (G3V4 Enhanced)? n
              Logged-In ACD Agents: 500            Logged-In Advocate Agents: 500
                                               Logged-In IP Softphone Agents: 500
```

Screen 248. Call Center Optional Features

VDN of Origin Announcement

Provides a short voice message to an agent indicating the city of origin of the caller or the service requested by the caller based on the VDN used to process the call.

VDN Return Destination

Allows an incoming trunk call to be placed back in vector processing after all parties, except the originator, drop.

VuStats

Allows you to present BCMS statistics on phone displays.

VDN Return Destination

Allows an incoming trunk call to be placed back in vector processing after all parties, except the originator, drop.

VuStats (G3V4 Enhanced)

Allows you to use the G3V4 VuStats enhancements including historical data and thresholds.

Logged-In ACD Agents

Number of ACD Agents contracted for. This field limits the number of logged-in ACD agents to a number no more than the maximum purchased. The value of this field indicates the total of ACD agents that can be logged-in simultaneously.

The limit applies to ACD agents on ACD and EAS calls. Auto-Available Split (AAS) agent ports are counted when they are assigned. AAS split or skill members are also counted. If the port for an AAS split/skill member is logged out, (for example, when a ringing call is redirected) the logged-in agent count is not updated. These counts are updated only during administration.

Logged-In Advocate Agents

Appears when the CentreVu Advocate field is **y**. Number of CentreVu Advocate Agents contracted for.

The total number of logged-in CentreVu Advocate agents must be equal to or less than the number allowed in the Logged-In ACD Agents field. The number of logged-in CentreVu Advocate agents counts towards the total number of logged-in ACD agents.

Logged-In IP Softphone Agents

Number of IP Softphone Agents contracted for. This field limits the number of logged-in IP Softphone agents to a number no more than the maximum purchased. The value of this field indicates the total of IP Softphone agents that can be logged-in simultaneously.

Field descriptions for QSIG Optional Features

```
                                          Page 7 of X
            QSIG OPTIONAL FEATURES

                                      Basic Call Setup? n
                        Basic Supplementary Services? n
                               Centralized Attendant? n
                               Interworking with DCS? n
                 Supplementary Services with Rerouting? n
                         Transfer into QSIG Voice Mail? n
                                  Value-Added (VALU)? n
```

Screen 249. QSIG Optional Features

Basic Call Setup

Provides basic QSIG services: basic connectivity and calling line ID number. To use this enhancement, either the ISDN-PRI or ISDN-BRI Trunks fields must be **y**.

Basic Supplementary Services

To use this enhancement, either the ISDN-PRI or ISDN-BRI Trunks fields must be **y**. Provides the following QSIG Supplementary Services:

- Name ID
- Transit Capabilities; that is, the ability to tandem QSIG information elements
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections
- Call Forwarding (Diversion) by forward switching. No reroute capabilities are provided
- Call Transfer by join. No path replacement capabilities are provided.
- Call Completion (also known as Automatic Callback)

Interworking with DCS

Allows the following features to work between a user on a DCS-enabled switch in a network and a QSIG-enabled switch:

- Called/calling party name display
- Called/calling party number display

This field cannot be set to **y** if the DCS (Basic) field is **n**.

Centralized Attendant

Can be enabled only if the Supplementary Services with Rerouting field is **y**. Allows all attendants in one location to serve users in multi locations. All signaling is done over QSIG ISDN lines.

Supplementary Services with Rerouting

Provides the following QSIG Supplementary Services:

- Transit Capabilities; that is, the ability to tandem QSIG information elements.
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections.
- Call Forwarding (Diversion) by forward switching. In addition, reroute capabilities are provided.
- Call Transfer by join. In addition, path replacement capabilities are provided.

Transfer Into QSIG Voice Mail

Can be enabled only if the Basic Supplementary Services field is **y** and either the ISDN-PRI Trunk or ISDN-BRI Trunk field is **y**. Allows transfer directly into the voicemail box on the voicemail system when a QSIG link connects Avaya MultiVantage and the voice mail.

Value Added

Provides additional QSIG functionality, including the ability to send and display calling party information during call alerting. See *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

Field descriptions for ASAI (page 8)

```
change system-parameters customer options                      Page X of X

                           ASAI FEATURES

                          CTI Stations? n
                         Phantom Calls? n



                      ASAI PROPRIETARY FEATURES

                          Agent States? n
```

Screen 250. ASAI Features

CTI Stations

Appears when the ASAI Link Plus Capabilities field is **y**. This field needs to be enabled for any application (using a link of Type ASAI) that uses a CTI station to receive calls.

For more information see the *Avaya MultiVantage CallVisor ASAI Technical Reference*.

Phantom Calls

Appears when the ASAI Link Plus Capabilities field is **y**. Enables phantom calls. The Phantom Calls field only applies to links administered as type ASAI.

For more information see the *Avaya MultiVantage CallVisor ASAI Technical Reference*.

Agent States

Appears when the Computer Telephony Adjunct Links field is **y**. The Agent States field provides proprietary information used by Avaya applications. For more information, contact your Avaya representative.

NOTE:
The Agent States field only applies to links administered as type adjlk. This field was previously named Proprietary Applications.

Field descriptions for Maximum IP Registrations by Product ID

```
                                              Page X of X
                     MAXIMUM IP REGISTRATIONS BY PRODUCT ID

Product ID_Rel. Limit     Product ID_Rel. Limit     Product ID_Rel. Limit
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
__________ ___. _____     __________ ___. _____     __________ ___. _____
```

Screen 251. Maximum IP Registrations by Product ID

Product ID

Identifies the product using the IP (internet protocol) registration.

Valid displays	Usage
IP_Phone	IP Telephones
IP_Soft	IP Softphones
IP_Agent	IP Agents
IP_eCons	SoftConsole IP attendant
IP_ROMax	R300 Remote Office phones

Rel

Release number of the IP endpoint.

Valid displays	Usage
0 to 99 or blank	Release number of the IP endpoint

Limit

Maximum number of IP registrations allowed.

Valid displays	Usage
1000 or 5000 depending on switch configuration	Maximum number of IP registrations allowed. For Avaya R300 Remote Office Communicator, defaults to the maximum allowed value for the Concurrently Registered Remote Office Stations on page 1 of this screen.

System Parameters OCM Call Classification

This screen enters the tone characteristics for your country for Outbound Call Management (OCM) applications. It is not required for United States OCM applications. If you cannot access this screen, contact your Avaya representative.

This screen appears when Global Call Classification field on the System Parameters Customer Options screen is set to y, or when the Enable Busy Tone Disconnect for Analog loop-start Trunks field on the System Parameters Country Options screen is set to y. This screen defines the busy tone and cadence and can be administered with up to 4 on and off steps, which is four valid cycles to determine busy tone.

We recommend that you use a minimum of two on and off steps to determine a valid busy tone. If the cadence is administered with one on and off step, any time the classifier hears the cadence it is considered BTD signal.

Field descriptions for page 1

```
                                                                  Page 1 of 2
                  SYSTEM PARAMETERS OCM CALL CLASSIFICATION


TONE DETECTION PARAMETERS
  Global Classifier Adjustment (dB): ___
              USA Default Algorithm? _n_
                  USA SIT Algorithm? ___
```

Screen 252. System Parameters OCM Call Classification screen

Global Classifier Adjustment (dB)

Enter a number to specify the dB loss adjustment.

Valid entries	Usage
0 to **15**	**0** is the least and **15** the most adjustment.

USA Default Algorithm

Valid entries	Usage
y/n	To use the default United States tone detection, set this field to **y**. If you enter **n**, the US Special Information Tones (SIT) Algorithm field appears.

USA SIT Algorithm

Valid entries	Usage
y	To use the United States (SIT) tone characteristics for SIT tone detection.
n	The system treats tones with the administered tone name “intercept” as if they were SIT VACANT, and treats tones with the administered tone name “information” as if they were SIT UNKNOWN.

Field descriptions for page 2

```
                                                              Page 2 of 9
                   SYSTEM PARAMETERS OCM CALL CLASSIFICATION

Tone Name   Instance       Tone       Cadence   Duration      Duration
                        Continuous    Step     Minimum       Maximum
_________    _____          ___       1.  on     _____         _____
                                      2.  off     _____         _____
                                      3.  on     _____         _____
                                      4.  off     _____         _____
                                      5.  on     _____         _____
                                      6.  off     _____         _____
                                      7.  on     _____         _____
                                      8.  off     _____         _____
```

Screen 253. System Parameters OCM Call Classification screen

Tone Name

This field is required for tone definition outside of the U.S. and Canada.

If the Global Call Classification field on the System Parameters Customer Options screen is **n**, only busy can be entered into this Tone Name field. If Busy Tone Disconnect is enabled, only busy can be entered into this field.

Valid entries	Usage
busy **information** **intercept** **reorder** **ringback**	Enter the name of the tone that you are adding or modifying. Enter **busy** for Busy Tone Disconnect.

Instance

Enter the instance number of the tone. If the system identifies a tone that matches the characteristics defined on more than one page of this screen the system applies the tone definition from the earlier page.

Valid entries	Usage
1–**8**	The number distinguishes tones that have the same name but more than one definition of silence and tone-on characteristics.

Tone Continuous

Valid entries	Usage
y	Indicates a continuous tone. If you enter **y**, you cannot enter data in the Duration Minimum or Duration Maximum fields.
n	Indicates a non-continuous tone.

Cadence Step

A display-only field identifying the number of each tone cadence step and indicating whether the tone is on or off during this cadence step.

Duration Minimum

Specifies the lower limit in milliseconds (msec) of the tone duration.

Valid entries	Usage
75 – **6375**	Enter in increments of 25 msec.

Duration Maximum

Specifies the upper limit in milliseconds of the tone duration.

NOTE:
On the Feature-Related System Parameters screen, set the Off-Premises Tone Detect Timeout Interval field to its maximum value.

Valid entries	Usage
75 – **6375**	Enter in increments of 25 msec.

Telecommuting Access

This screen allows the System Administrator to administer the extension which allows remote users to use the feature.

Field descriptions for page 1

```
add telecommuting-access
                              TELECOMMUTING ACCESS

                   Telecommuting Access Extension: ________
```

Screen 254. Telecommuting Access

Telecommuting Access Extension

This only allows remote access to the Telecommuting Access feature.

Valid entries	Usage
Unassigned extension of 1-7 digits	Enter an extension that conforms to your system's dial plan and is not assigned to any other system object.

Related Topics

Refer to ''Configuring Avaya MultiVantage for telecommuting'' on page 351 for information about setting up telecommuting.

Tenant

This screen defines tenants to the system. If your switch uses tenant partitioning, see “Tenant Partitioning” on page 1749 for more information.

Field descriptions for page 1

```
change tenant 18
                                                    Tenant 18

                             Tenant Description: ________________________________________

                                Attendant Group: 1

                          Ext Alert Port (TAAS): _______ Ext Alert (TAAS) Extension: ____

                              Night Destination: _____

                                   Music Source: 1

                       Attendant Vectoring VDN:

                               Emergency Number:
```

Screen 255. Tenant screen

Tenant

This is a display only field. It contains the tenant number that you entered on the command line.

Tenant Description

Valid entries	Usage
40 alpha-numeric characters or blank	You may leave the description field blank, but future administration will be easier if you provide descriptive information.

Attendant Group

This required information relates a tenant to an attendant group.

NOTE:
The default for the system is that all attendant groups exist. However, the attendant group will be empty if no consoles are assigned to it.

Valid entries	Usage
1–**16**.	See *Avaya MultiVantage™ Hardware Solutions Guide*, for your system's range of allowable attendant group numbers.

Ext Alert Port (TAAS)

Valid entries	Usage
A valid port address or X	Enter 7 character port number. Enter Trunk Answer Any Station (TAAS) alert port information, if any. The port type and the object type must be consistent, and the port can be assigned to only one tenant.

Ext Alert (TAAS) Extension

This field appears only if you entered an **X** in Ext Alert Port (TAAS).

Valid entries	Usage
A valid extension	If you entered an **X** in Ext Alert Port (TAAS), you must enter extension information in this field.

Night Destination

Valid entries	Usage
A valid extension	Enter the night service station extension, if you want night service for this tenant.

Music Source

Valid entries	Usage
1–**20**	Enter the music/tone source for this partition. These sources are defined on the Music Sources screen.

Attendant Vectoring VDN

This field appears only if, on the System-Parameters Customer-Options screen, the Attendant Vectoring field is **y** and the Tenant Partitioning field is **n**. Enter the assigned Attendant VDN extension. When set to **y**, the VDN and Vector screens display.

Emergency Number

Enter the phone number you want to use for this tenant's emergency calls. If your system is in a No-License mode, these will be the only numbers that the tenant can dial. The number may contain the feature access code for Emergency Access to the Attendant, trunk access codes, or any number, *, or #.

Field descriptions for page 2

```
change tenant 18
                                   Tenant 18


CALLING PERMISSION (Enter y to grant permission to call specified Tenant)


 1? y    11? n    21? n    31? n    41? n    51? n    61? n    71? n    81? n    91? n
 2? n    12? n    22? n    32? n    42? n    52? n    62? n    72? n    82? n    92? n
 3? n    13? n    23? n    33? n    43? n    53? n    63? n    73? n    83? n    93? n
 4? n    14? n    24? n    34? n    44? n    54? n    64? n    74? n    84? n    94? n
 5? n    15? n    25? n    35? n    45? n    55? n    65? n    75? n    85? n    95? n
 6? n    16? n    26? n    36? n    46? n    56? n    66? n    76? n    86? n    96? n
 7? n    17? n    27? n    37? n    47? n    57? n    67? n    77? n    87? n    97? n
 8? n    18? y    28? n    38? n    48? n    58? n    68? n    78? n    88? n    98? n
 9? n    19? n    29? n    39? n    49? n    59? n    69? n    79? n    89? n    99? n
10? n    20? n    30? n    40? n    50? n    60? n    70? n    80? n    90? n   100? n
```

Screen 256. Tenant

Tenant

This is a display only field. It contains the tenant number that you entered on the command line.

Calling permissions

The system default allows each tenant to call only itself and Tenant 1. If you want to change that, you can do that on this screen.

Valid entries	Usage
y/n	Enter **y** to establish calling permission between the tenant number that you entered on the command line and any other tenant.

Terminal Parameters

This screen administers system-level parameters and audio levels for the 603 CALLMASTER telephones and the 4600-series, 6400-series, 8403, 8405B, 8405B+, 8405D, 8405D+, 8410B, 8410D, 8411B, 8411D, and 8434D phones.

NOTE:
Only authorized Avaya personnel can administer this screen.

```
change terminal-parameters 8400                                     Page 1 of 1
                                8400-TYPE TERMINAL PARAMETERS

         Default Parameter Set: __          Customize Parameters? _

   OPTIONS*
                     Display Mode: _*               DLI Voltage Level: _________*
     Handset Expander Enabled? _*

   PRIMARY LEVELS*
            Voice Transmit (dB): _____*         Voice Sidetone (dB): _____*
             Voice Receive (dB): _____*     Touch Tone Sidetone (dB): _____*
       Touch Tone Transmit (dB): _____*

   ADJUNCT LEVELS*+
            Voice Transmit (dB): _____*         Voice Receive (dB): _____*
            Voice Sidetone (dB): _____*    Touch Tone Sidetone (dB): _____*
   BUILT-IN SPEAKER LEVELS
            Voice Transmit (dB): _____*         Voice Receive (dB): _____*
                                           Touch Tone Sidetone (dB): _____*

   8403 BUILT-IN SPEAKER LEVELS
             Voice Receive (dB): _____*    Touch Tone Sidetone (dB): _____*
```

Screen 257. 8400-Series Terminal Parameters

* This field appears only if Customize Parameters is **y**.

```
change terminal-parameters 302/603                             Page 1 of 1
                                   302/603 TERMINAL PARAMETERS

         Default Parameter Set: __              Customize Parameters? _

   OPTIONS*
                     Display Mode: _*               DLI Voltage Level: _________*


   PRIMARY LEVELS*
            Voice Transmit (dB): _____*         Voice Sidetone (dB): _____*
             Voice Receive (dB): _____*     Touch Tone Sidetone (dB): _____*
       Touch Tone Transmit (dB): _____*
```

Screen 258. 603/302 Terminal Parameters

```
change terminal-parameters 6400/607A1/4600/2420                      Page 1 of 1

                    6400/607A1/4600/2420-TYPE TERMINAL PARAMETERS

          Default Parameter Set: 1         Customize Parameters? y
   Note: LEVELS do not apply to the 4600 terminals.
    OPTIONS
                    Display Mode:         Handset Expander Enabled?
           Volume for DCP Types:
            Volume for IP Types:
    PRIMARY LEVELS
             Voice Transmit (dB):              Voice Sidetone (dB):
              Voice Receive (dB):         Touch Tone Sidetone (dB):
        Touch Tone Transmit (dB):

    BUILT-IN SPEAKER LEVELS
             Voice Transmit (dB):               Voice Receive (dB):
                                          Touch Tone Sidetone (dB):
    6402 BUILT-IN SPEAKER LEVELS
              Voice Receive (dB):         Touch Tone Sidetone (dB):
```

Screen 259. 6400/607A1/4600 Type Terminal Parameters

* This field appears only if Customize Parameters is **y**

Default Parameter Set

Determines which default set of telephone options and levels will be used. This field corresponds to the country codes. Refer to ''System Parameters Country-Options'' on page 1111 for the country code listing.

Customize Parameters

Indicates whether the administrator wishes to change one or more of the default parameters.

Valid entries	Usage
y	If this field is **y** (yes), the OPTION and LEVEL fields appear and all fields can be edited.
n	If this field is **n** (no), the system uses all default parameters associated with the Default Parameter Set field and all fields are display-only.

Display Mode

Determines how the #) and ~ characters appear on the phone's display.This field only appears if Customize Parameters is **y**.

Valid entries	Usage
1	If this field is set to **1**, the # and ~ do not change.
2	If this field is set to **2**, the phone displays a # as a British pound sterling symbol and a ~ as a straight overbar.

DLI Voltage Level

Determines whether DCP Line Voltage used by the telephones is forced high, forced low, or allowed to automatically adjust. This field only appears if Customize Parameters is **y**.

Handset Expander Enabled

Determines whether the telephone will reduce noise on the handset. This field appears only if Customize Parameters is **y**.

Valid entries	Usage
y	If the field is **y**, the telephone reduces background noise.

Primary levels

The following fields only appear if Customize Parameters is set to **y**. In each case, if the field is blank, the system uses the default setting from the Default Parameter Set. Also, these fields all require the same input; valid entries are from **-44.0** db through **+14.0** db in 0.5 increments (for example, -44.0, -43.5, -43.0 and son on).

Volume for DCP Types

This field allows the DCP phone volume to be adjusted while the call is in progress.

Valid entries	Usage
default speaker, handset unchangeable	The speaker resets to the default settings while the adjusted handset setting is retained.
default settings used to begin each call	No adjusted handset and speaker settings are retained.
retain handset and speaker between calls	The adjusted handset and speaker settings are retained.
retain speaker, handset unchangeable	Only the adjusted speaker setting is retained.

Volume for IP Types

This field allows the IP phone volume to be adjusted while the call is in progress.

NOTE:
If you use this field, Avaya recommends that you not change any values in the PRIMARY LEVELS or BUILT-IN SPEAKER LEVELS areas.

Valid entries	Usage
default speaker, handset unchangeable	The speaker resets to the default settings while the adjusted handset setting is retained.
default settings used to begin each call	No adjusted handset and speaker settings are retained.
retain handset and speaker between calls	The adjusted handset and speaker settings are retained.
retain speaker, handset unchangeable	Only the adjusted speaker setting is retained.

Voice Transmit (dB)

Determines the volume of voice outbound from the telephone.

Voice Receive (dB)

Determines the volume of voice inbound to the telephone.

Voice Sidetone (dB)

Determines the volume of voice fed back from the handset voice microphone to the user’s ear.

Touch Tone Sidetone (dB)

Determines the touchtone volume fed back from the telephone when a users presses a button.

Touch Tone Transmit (dB) —

Determines the touchtone volume fed outbound from the telephone.

NOTE:
You cannot administer all five of the Primary Level fields to +14.0 dB. If you attempt to submit the Terminal Parameters screen with all Primary Levels set to +14.0 dB, you receive an error message.

Adjunct levels

The following fields appear only if you are administering 8400-series telephones and if Customize Parameters is **y**. In each case, if the field is blank, the system uses the default setting from the Default Parameter Set. Also, these fields all require the same input; valid input is listed in “Primary levels” above.

Voice Transmit (dB)

Determines the volume of voice outbound from the adjunct.

Voice Receive (dB)

Determines the volume of voice inbound to the adjunct.

Voice Sidetone (dB)

Determines the volume of voice fed back from the handset voice microphone to the user’s ear.

Touch Tone Sidetone (dB)

Determines the touchtone volume fed back from the telephone when a users presses a button.

Terminating Extension Group

This screen defines a Terminating Extension Group (TEG). Any phone can be assigned as a TEG member; however, only a multi-appearance phone can be assigned a TEG button with associated status lamp. The TEG button allows the phone user to select a TEG call appearance for answering or for bridging onto an existing call.

The TEG members are assigned on an extension number basis. Call reception restrictions applicable to the group are specified by the group class of restriction (COR). The group COR takes precedence over an individual member’s COR. The members could all be termination restricted but still receive calls if the group is not restricted.

The system allows for as many as 32 TEGs with up to 4 members each. An extension number can be assigned to more than one TEG but can have only one appearance of each group.

Field descriptions for page 1

```
change term-ext-group 1                                       Page   1 of   1
                       TERMINATING EXTENSION GROUP

    Group Number: 1                            Group Extension:
      Group Name:                                Coverage Path:
   Security Code:                                          COR: 1
                                                            TN: 1
ISDN Caller Disp:                                    LWC Reception: none
      AUDIX Name:                        Messaging Server Name:

GROUP MEMBER ASSIGNMENTS

    Ext      Name (first 26 characters)    Ext       Name (first 26 characters)
 1: 51001    26 character name sta 51001   3:
 2:                                        4: 51002   26 character name sta 51002
```

Screen 260. Terminating Extension Group

Group Number

A display-only field when the screen is accessed using an administration command such as **add** or **change**.

Group Extension

Enter the extension of the terminating extension group.

Valid entries	Usage
1 to 7 digits	Unused extension number (may not be a VDN extension). Do not leave blank.

Group Name

Enter the name used to identify the terminating extension group.

Coverage Path

Enter a number for the call coverage path for this group. A TEG cannot serve as a coverage point; however, calls to a TEG can redirect to coverage.

Valid entries	Usage
1 to **999**	
t1 to **t999**	time of day table

Security Code

Valid entries	Usage
4-digit security code	This code is used for the Demand Print feature.

COR

Valid entries	Usage
0 through **95**	Enter the class of restriction (COR) number that reflects the desired restrictions.

TN

Valid entries	Usage
1 to **20**	Enter the Tenant Partition number.

ISDN Caller Disp

This field is required if, on the System-Parameters Customer-Options screen, the ISDN-PRI or ISDN-BRI Trunks field is **y**.

Valid entries	Usage
grp-name	Specify whether the TEG group name or member name (member of TEG where call terminated) will be sent to the originating user.
mbr-name	Specify whether the TEG group name or member name (member of TEG where call terminated) will be sent to the originating user.
blank	If the ISDN-PRI or ISDN-BRI Trunks field is **n**, leave blank.

LWC Reception

Defines the source for Leave Word Calling (LWC) messages.

Valid entries	Usage
audix	For G3i systems, if LWC is attempted, the messages are stored in AUDIX.
msa-spe	For G3i systems, if LWC is attempted, the messages are stored in the system processing element (spe).
none	For G3i systems, if LWC is attempted, the messages are not stored.

Valid entries	Usage
audix	For DEFINITY Server R systems, if LWC is attempted, the messages are stored in AUDIX.
msa	For DEFINITY Server R systems, if LWC is attempted, the messages are stored in the Message Server Adjunct - Switch Processor.
spe	For DEFINITY Server R systems, if LWC is attempted, the messages are stored in the system processing element (spe).
none	For DEFINITY Server R systems, if LWC is attempted, the messages are not stored.

AUDIX Name

Name of the AUDIX machine as it appears in the Node Names screen. Only appears for an DEFINITY Server R configuration.

Valid entries	Usage
	Unique identifiers for adjunct equipment.

Messaging Server Name

Name of the server as it appears in the Node Names screen. Only appears for DEFINITY Server R configurations.

Valid entries	Usage
	Unique identifiers for messaging server equipment.

Group Member Assignments — Ext

Enter the extension number (may not be a VDN extension) assigned to a station.

Valid entries	Usage
	An extension number of 1-7 digits.

Group Member Assignments — Name

This display-only field shows the name assigned to the preceding extension number when the TEG member’s phone is administered.

TFTP Server

The Trivial File Transfer Protocol screen allows specification of the TFTP server the Avaya Call Processing engine uses to get download files.

```
change tftp-server                                             Page 1 of 1
                        TFTP Server Configuration

      Local Node Name:
TFTP Server Node Name:
     TFTP Server Port: 69
     File to Retrieve:

          File Status:
            File Size:
   Filename in Memory:
```

Screen 261. TFTP Server screen

Local Node Name

The local node name must be a valid entry from the IP Node Names screen. The node must be assigned to a CLAN ip-interface or **procr** (processor CLAN).

Valid entries	Usage
1-15 characters	
procr	Processor CLAN for Avaya S8300 Media Server, Avaya S8700 Multi-Connected Media Server, and Avaya S8700 IP Connected Media Server.

TFTP Server Node Name

The TFTP server node name must be a valid entry from the IP Node Names screen.

Valid entries	Usage
1-15 characters	

TFTP Server Port

Valid entries	Usage
1-64,500	Enter a number for the remote TCP port.

File to Retrieve

Valid entries	Usage
	Enter the name of the file you are going to retrieve using up to 32 alpha-numeric, case sensitive, characters for identification.

File Status

A display-only field showing Download In Progress, Download Failed, File Not Found, or Download Completed.

File Size

A display-only field showing the number of bytes transferred.

Filename in Memory

A display-only field showing the name of the file currently in ACP memory.

Time of Day Coverage Table

This screen allows up to five different coverage paths, associated with five different time ranges, for each day of the week. Only one coverage path can be in effect at any one time.

Field descriptions for page 1

```
change coverage time-of-day 3
                    TIME OF DAY COVERAGE TABLE 3___

           Act   CVG      Act   CVG      Act   CVG      Act   CVG      Act   CVG
           Time  PATH     Time  PATH     Time  PATH     Time  PATH     Time  PATH

  Sun      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
  Mon      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
  Tue      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
  Wed      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
  Thu      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
  Fri      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
  Sat      00:00 ____     __:__ ____     __:__ ____     __:__ ____     __:__ ____
```

Screen 262. Time of Day Coverage Table

Time of Day Coverage Table

A display-only field when the screen is accessed using an administration command. Specifies the Time of Day Coverage Table number. Up to 999 can be administered.

Act Time

Specify the activation time of the coverage path administered in the next CVG PATH field. Enter the information in 24-hour time format.

Valid entries	Usage
00:01 to **23:59**	If there are time gaps in the table, there will be no coverage path in effect during those periods. The first activation time for a day is set to 00:00 and cannot be changed. Activation times for a day must be in ascending order from left to right.

CVG Path

Enter the coverage path number.

Valid entries	Usage
1 through **9999**	For the DEFINITY Server R configurations
1 through **999**	For the DEFINITY Server SI configurations

Time of Day Routing Plan

Use this screen to set up Time of Day Routing Plans. You can route AAR and ARS calls based on the time of day each call is made. You can design up to 8 Time of Day Routing Plans, each scheduled to change up to 6 times a day for each day in the week.

Match the Time of Day Routing Plan PGN# with the PGN# field on the Partition Routing Table for the route pattern you want to use.

NOTE:
Automatic Route Selection (ARS) or Private Networking, AAR/ARS Partitioning, and Time of Day Routing must be enabled on the

System-Parameters Customer-Options screen before you can use Time of Day Routing.

```
change time-of-day
                                   TIME OF DAY ROUTING PLAN ___                Page 1 of 1
       Act     PGN    Act     PGN    Act      PGN    Act      PGN    Act      PGN    Act      PGN
       Time     #     Time     #     Time      #     Time      #     Time      #     Time      #
Sun   00:00    1    __:__    _     __:__     _     __:__     _     __:__     _     __:__     _
Mon   00:00    1    __:__    _     __:__     _     __:__     _     __:__     _     __:__     _
Tue   00:00    1    __:__    _     __:__     _     __:__     _     __:__     _     __:__     _
Wed   00:00    1    __:__    _     __:__     _     __:__     _     __:__     _     __:__     _
Thu   00:00    1    __:__    _     __:__     _     __:__     _     __:__     _     __:__     _
Fri   00:00    1    __:__    _     __:__     _     __:__     _     __:__     _     __:__     _
Sat   00:00    1    __:__    _     __:__     _     __:__     _     __:__     _     __:__     _
```

Screen 263. Time Of Day Routing Plan

Time of Day Routing Plan

Displays the Time of Day Routing Plan number (1 through 8).

Act Time

Specifies the time of day the route pattern (identified by PGN) begins.

Valid entries	Usage
00:00 to **23:59**	Time is represented using a 24 hour clock. List times for the same day in increasing order. There must be at least one entry for each day.

PGN

Identifies the route pattern for activation time listed.

Valid entries	Usage
1 to **8**	Enter a PGN that matches the PGN and route pattern on the Partition Routing Table. There must be at least one entry for each day.

Toll Analysis

NOTE:
The Toll List field on this screen does not interact with or relate to the ARS Toll Table.

This screen associates dialed strings to the system’s Restricted Call List (RCL), Unrestricted Call List (UCL), and Toll List. You can force users to dial an account code if you associate dialed strings with CDR Forced Entry of Account Codes.

To maximize System security, it is recommended that toll calling areas be restricted as much as possible through the use of the RCL (Restricted Call List) and Toll List fields on this screen.

```
change toll analysis                                              Page 1 of 1
                                TOLL ANALYSIS                  Percent Full:_
                               Location:
                         Total        Toll  CDR  <--Unrestricted Call List-->
    Dialed String      Min  Max  RCL  List  FEAC  1  2  3  4  5  6  7  8  9 10
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
_____________________  ___  ___   _     _     _   _  _  _  _  _  _  _  _  _  _
```

Screen 264. Toll Analysis

Percent Full

Displays the percentage (**0 to 100**) of the system's memory resources that have been used by AAR/ARS. If the figure is close to 100%, you can free-up memory resources.

Location

Display only field. Values other than "all" appear if the Multiple Locations field on the System Parameters Customer Options is **y**.

Valid display	Usage
1 to **44**	Defines the switch location for this Toll Analysis Table.
all	Indicates that this Toll Analysis Table is the default for all port network (cabinet) locations.

Dialed String

Valid entries	Usage
digits **0** to **9** (up to 18 characters)	Enter the dialed string you want the switch to analyze.
***, x, X**	wildcard characters

Min

Valid entries	Usage
1 to Max	Enter the minimum number of user-dialed digits the system collects to match to this dialed string.

Max

Valid entries	Usage
Min to **24**	Enter the maximum number of user-dialed digits the system collects to match to this dialed string.

RCL

Enter **x** to assign the Dialed String to the Restricted Call List (RCL).

Valid entries	Usage
x	All entries of **x** and their associated dialed strings are referred to as the System's Restricted Call List. The RCL can be assigned to any COR. A call attempt from a facility whose COR is marked as being associated with the RCL and whose dialed string matches a RCL dialed string field will be denied. The caller receives intercept treatment.

Toll List

Valid entries	Usage
x	Enter **x** to assign the Dialed String to the Toll List.

Dialed String	Min	Max	Toll List
0	1	23	x
1	4	23	x
20	10	10	x
21	10	10	x
30	10	10	x
31	10	10	x
40	10	10	x
41	10	10	x
50	10	10	x

Dialed String	Min	Max	Toll List
51	10	10	x
60	10	10	x
61	10	10	x
70	10	10	x
71	10	10	x
80	10	10	x
81	10	10	x
90	10	10	x
91	10	10	x

CDR FEAC

Valid entries	Usage
x	Enter **x** to require an account code from a call whose facility COR requires a Forced Entry of Account Code.

Unrestricted Call List

Valid entries	Usage
x	Enter **x** to assign the dialed string to one of the system's Unrestricted Call Lists (UCL).

Trunk Group

Use the Trunk Group screen to set basic characteristics for every type of trunk group and to assign ports to the group. This section lists and describes all the fields you might see on the screen. Many fields are dependent on the settings of other fields and only appear when certain values are entered in other fields on the screen. For example, the entry in the Group Type field may significantly change the content and appearance of the Trunk Group screen.

For descriptions of the unique fields used with ISDN trunks, see ''ISDN trunk group'' on page 894.

Field descriptions for page 1

The figure below shows a common configuration for page 1 of the Trunk Group screen when the Group Type field is **tie**. This screen is only an example, and the fields shown below may change or disappear according to specific field settings.

```
add trunk-group next                                                  Page  1 of  x
                                  TRUNK GROUP

 Group Number: ___                   Group Type: ________        CDR Reports: _
   Group Name: ___________________________ COR: __        TN: ___        TAC: ____
    Direction: ________        Outgoing Display? _    Trunk Signaling Type: ____
  Dial Access? _                 Busy Threshold: ___         Night Service: ________
 Queue Length: ___                                    Incoming Destination: ________
    Comm Type: _____                 Auth Code? _
                                   Trunk Flash? _
          BCC: _                           ITC? ____
 TRUNK PARAMETERS
    Trunk Type (in/out): ____________      Incoming Rotary Timeout(sec): __
     Outgoing Dial Type: _________                   Incoming Dial Type: ________
                                                Disconnect Timing(msec): ____
        Digit Treatment: __________                              Digits: ____
                                                      Sig Bit Inversion: none
      Analog Loss Group: ___                         Digital Loss Group: ___
     Incoming Dial Tone? _
               Bit Rate: _____       Synchronization: _____      Duplex: ____
  Disconnect Supervision - In? _  Out? _
  Answer Supervision Timeout: ___        Receive Answer Supervision? _
```

Screen 265. Tie Trunk Group

Group Number

This field displays the group number assigned when the trunk group was added.

NOTE:
For trunk groups connecting 2 switches in Distributed Communication System networks, Avaya recommends that you assign the same group number on both switches.

Group Type

Enter the type of trunk group.

Busy-out the trunk group before you change the group type. Release the trunk group after you make the change. For more information about busying out and releasing trunk groups, refer to your system's Maintenance manual.

Valid entries	Usage
Access	Use access trunks to connect satellite switches to the main switch in Electronic Tandem Networks (ETN). Access trunks do not carry traveling class marks (TCM) and thus allow satellite callers unrestricted access to out-dial trunks on the main switch. Allows Inband ANI.
APLT	Advanced Private Line Termination (APLT) trunks are used in private networks. APLT trunks allow inband ANI.
CAMA	CAMA trunks route emergency calls to the local community's Enhanced 911 systems.
CO	CO trunks typically connect your switch to the local central office, but they can also connect adjuncts such as external paging systems and data modules.
CPE	Use CPE trunks to connect adjuncts, such as paging systems and announcement or music sources, to the switch.
DID	Use DID trunks when you want people calling your organization to dial individual users directly without going through an attendant or some other central point. Allows Inband ANI.
DIOD	DIOD trunks are two-way trunks that transmit dialed digits in both directions. In North America, use tie trunks for applications that require two-way transmission of dialed digits. Allows Inband ANI.
DMI-BOS	Digital Multiplexed Interface – Bit-Oriented Signaling (DMI-BOS) trunks allow communication with systems using DMI-BOS protocol. DMI-BOS trunks allow inband ANI.
FX	An FX trunk is essentially a CO trunk that connects your switch directly to a central office outside your local exchange area. Use FX trunks to reduce long distance charges if your organization averages a high volume of long-distance calls to a specific area code.

Valid entries	Usage
ISDN	Use ISDN trunks when you need digital trunks that can integrate voice, data, and video signals and provide the bandwidth needed for applications such as high-speed data transfer and video conferencing. ISDN trunks can also efficiently combine multiple services on one trunk group. Use ISDN for Network Call Transfer. For more information, see *Avaya MultiVantage Network Call Redirection* (555-233-759). You cannot enter isdn unless the ISDN-PRI field, the ISDN-BRI Trunks field, or both have been enabled on the System-Parameters Customer-Options screen.
RLT	Release Link trunks work with Centralized Attendant Service in a private network.
Tandem	Tandem trunks connect tandem nodes in a private network. Tandem trunks allow inband ANI.
Tie	Use tie trunks to connect a switch to a central office or to another switch in a private network. Tie trunks transmit dialed digits with both outgoing and incoming calls, and allow inband ANI.
WATS	Use WATS trunks to reduce long-distance bills when your organization regularly places many calls to a specific geographical area in North America. Outgoing WATS service allows calls to certain areas (“WATS bands”) for a flat monthly charge. Incoming WATS trunks allow you to offer toll-free calling to customers and employees.

CDR Reports

Valid entries	Usage
y	All outgoing calls on this trunk group will generate call detail records. If the Record Outgoing Calls Only field on the CDR System Parameters screen is **n**, then incoming calls on this trunk group will also generate call detail records.
n	Calls over this trunk group will not generate call detail records.
r (ring-intvl)	CDR records will be generated for both incoming and outgoing calls. In addition, the following ringing interval CDR records are generated: ■ Abandoned calls: The system creates a record with a condition code of “H,” indicating the time until the call was abandoned. ■ Answered calls: The system creates a record with a condition code of “G,” indicating the interval from start of ring to answer. ■ Calls to busy stations: The system creates a record with a condition code of “I,” indicating a recorded interval of 0.

NOTE:
For ISDN trunk groups, the Charge Advice field affects CDR information. For CO, DIOD, FX, and WATS trunk groups, the Analog PPM field affects CDR information.

Group Name

Valid entries	Usage
1 to 27 characters	Enter a unique name that provides information about the trunk group. Don’t use the default entry or the group type (DID, WATS) here. For example, you might use names that identify the vendor and function of the trunk group: USWest Local; Sprint Toll, etc.

COR

Valid entries	Usage
0 to **95**	Enter a class of restriction (COR). Classes of restriction control access to trunk groups, including trunk-to-trunk transfers.

Tip:
Remember that facility restriction levels (FRL) are assigned to classes of restriction. Even if 2 trunk groups have classes of restriction that allow a connection, different facility restriction levels may prevent operations such as off-net call forwarding or outgoing calls by remote access users.

TN

Valid entries	Usage
1 to **20** (DEFINITY Server SI / DEFINITY Server CSI)	Enter a Tenant Partition number to assign this trunk group to the partition.
1 to **100** (DEFINITY Server R)	

Tip:
Double-check your entry. If you accidentally enter an unassigned tenant partition number, the system will accept the entry but no calls will go to the trunk group.

TAC

Enter the trunk access code (TAC) that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. CDR reports use the TAC to identify each trunk group.

Valid entries	Usage
1-4 digit number	Enter any number that fits the format for trunk access codes or dial access codes defined in your dial plan.
***, #**	* and # may be used as the first character in a TAC.

CO Type

This field appears when the Country field is **14** and is used only by trunk group members administered on a TN464D vintage 2 or later DS1 circuit pack.

Valid entries	Usage
analog **digital**	This field specifies whether the trunk group is connected to analog or digital facilities at the central office.

Protocol Type

This field specifies the type of line signaling protocol used for DID and DIOD trunk groups. This field appears when the Country field is **15** and is used only by trunk group members administered on a TN2199 or TN464D vintage 3 or later circuit pack.

Valid entries	Usage
inloc (Incoming local) **intol** (Incoming toll)	Enter the protocol the central office is using for this trunk group. Only the inloc protocol provides ANI.

Direction

Enter the direction of the traffic on this trunk group. The entry in this field affects which timers appear on the Administrable Timers page. This field appears for all trunk groups except DID and CPE.

Valid entries	Usage
incoming	
outgoing	
two-way	Enter two-way for Network Call Redirection.

Outgoing Display

This field allows display phones to show the name and number of the trunk group used for an outgoing call before the call is connected. This information may be useful to you when you’re trying to diagnose trunking problems.

Valid entries	Usage
y	Displays the trunk group name and number.
n	Displays the digits the caller dials.

CESID I Digits Sent

For emergency 911 service, your switch may send Caller's Emergency Service Identification (CESID) information to the central office or E911 tandem switch. This digit string is part of the E911 signaling protocol. This field appears when Group Type is cama.

Valid entries	Usage
1 to 3 digits	Determine the correct entry for this field by talking to your E911 provider.

Trunk Signaling Type

This field controls the signaling used by members in private network trunk groups, mainly in Italy, Brazil, and Hungary. This field appears if the Group Type is **access, aplt, diod, rlt, tandem**, or **tie**. Entries in this field affect which timers appear on the Administrable Timers page.

This field also controls the signaling used by members in public network digital trunk groups. This field displays if the Group Type field is **access**, **aplt**, **rlt**, **tandem**, or **tie**. Entries in this field affect which timers appear on the Administrable Timers page.

Valid entries	Usage
cont (continuous) **dis** (discontinuous)	E&M trunks in Italy, Brazil, and Hungary can use either continuous or discontinuous signaling. Each entry specifies a set of signals and available timers used in the process of setting up and releasing connections. The type of signaling you select on your switch must match the signaling type administered on the far-end switch. Use these values only when all trunk group members are assigned to ports on a TN464F, TN2464, or TN2140 circuit pack. Entering one of these values causes the Send Release Ack, Receive Release Ack, and Send Answer Supervision fields to appear. Refer to ''Trunk Type (in/out)'' on page 1182 for more information.
blank	If this field is blank, ports from any other tie trunk circuit pack may be added as trunk group members.

Use the following entries for tie trunks in Main-Satellite/Tributary networks. Each entry defines a function of the trunk group in the network. Use these values only when all trunk group members are assigned to a TN497 circuit pack.

tgu (for outgoing trunks)	Enter **tgu** at the main switch to administer a tie trunk group connected to a satellite switch. (This same group should be administered as **tge** at the satellite.)

Valid entries	Usage
tge (for incoming trunks)	Enter **tge** at a satellite switch to administer a tie trunk group connected to the main switch. (This same group should be administered as **tgu** at the main switch.)
tgi (for internal trunks)	Enter **tgi** at to administer a two-way tie trunk group between 2 satellites or between the main switch and a satellite. (This trunk group should be administered as **tgi** on both switches.)

DIOD trunks support pulsed and continuous E&M signaling in Brazil and discontinuous E&M signaling in Hungary. Use the following entries for DIOD trunks. Use these values only when all trunk group members are assigned to a TN464F (or later version) or TN2464 circuit pack.

cont	Enter **cont** for continuous E&M signaling**.**
pulsed	Enter **pulsed** for pulsed E&M signaling.
discont	Leave blank forR2 signaling.
	Hungary uses discontinuous E&M signaling when this field is **dis**. Brazil E&M trunks use continuous and pulsed E&M.

Dial Access

This field controls whether users can route outgoing calls through an outgoing or two-way trunk group by dialing its trunk access code. Allowing dial access does not interfere with the operation of AAR/ARS.

> ⚠ **SECURITY ALERT:**
> *Calls dialed with a trunk access code over WATS trunks bypass AAR/ARS and aren't restricted by facility restriction levels. For security, you may want to leave the field set to n unless you need dial access to test the trunk group.*

Valid entries	Usage
y	Allows users to access the trunk group by dialing its access code.
n	Does not allow users to access the trunk group by dialing its access code. Attendants can still select this trunk group with a Trunk Group Select button.

Busy Threshold

Use this field if you want attendants to control access to outgoing and two-way trunk groups during periods of high use. When the threshold is reached and the warning lamp for that trunk group lights, the attendant can activate trunk group control: internal callers who dial out using a trunk access code will be connected to the attendant, and the attendant can prioritize outgoing calls for the last remaining trunks. Calls handled by AAR and ARS route patterns go out normally.

Valid entries	Usage
1 to **99** **(**DEFINITY Server SI / DEFINITY Server CSI)	Enter the number of trunks that must be busy in order to light the warning lamp on the Attendant Console. For example, if there are 30 trunks in the group and you want to alert the attendant whenever 25 or more are in use, enter **25**.
1 to **255** (DEFINITY Server R)	

Night Service

This field sets the destination to which incoming calls go when Night Service is in operation. If a Night field on the Group Member Assignments page is administered with a different destination, that entry will override the group destination for that trunk. CPE, DID, and DIOD trunk groups do not support night service.

Tip:
Whenever possible, use a night service destination on your switch: otherwise some features won't work correctly, even over a DCS network.

Valid entries	Usage
blank	Leave this field blank if the Trunk Type (in/out) field is not auto/….
An extension number (can be a VDN)	Enter the extension of your night service destination.
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.

Queue Length

Outgoing calls can wait in a queue, in the order in which they were made, when all trunks in a trunk group are busy. If you enter 0, callers receive a busy signal when no trunks are available. If you enter a higher number, a caller hears confirmation tone when no trunk is available for the outgoing call. The caller can then hang up and wait: when a trunk becomes available, the switch will call the extension that placed the original call. The caller will hear 3 short, quick rings. The caller doesn't need to do anything but pick up the handset and wait: the switch remembers the number the caller dialed and automatically completes the call.

This field appears when the Direction field is **outgoing** or **two-way**.

Valid entries	Usage
0 to **100**	Enter the number of outgoing calls that you want to be held waiting when all trunks are busy. Enter **0** for DCS trunks.

Country

This field is administered at installation and sets numerous parameters to appropriate values for the public network in which the switch operates. For example, the value of this field, with the values of the Trunk Termination and the Trunk Gain fields, determines the input and trans-hybrid balance impedance requirements for ports on TN465B, TN2146, and TN2147 circuit packs.

This field appears for the trunk groups that connect Avaya MultiVantage to a central office in the public network — CO, DID, DIOD, FX, and WATS trunk groups.

CAUTION:
Don't change this field. If you have questions, contact your Avaya representative.

Valid entries	Usage
1 to **25**	Set at installation.
11	If the Country field is 11, Avaya MultiVantage is administered for Public Network Call Priority (Call Retention and Rering).
14	If the Country field is 14 and the Group Type is DID or DIOD, the CO Type field appears.
15	If the Country field is 15, Avaya MultiVantage is administered for Public Network Call Priority (Intrusion and Rering). Also, the Protocol Type field appears for Group Type DID or DIOD.
18	If the Country field is 18, Avaya MultiVantage can be administered for Public Network Call Priority (Mode of Release Control, Forced Disconnect, and Rering).
23	If the Country field is 23 and the Group Type field is either CO or DID, Avaya MultiVantage is administered for Block Collect Calls.

Version

Use this field to adjust the signaling on multi-country CO trunk circuit packs. Entries in this field adjust signaling characteristics on these circuit packs to match the signaling characteristics of the public network in a specific country. The field appears only for CO, FX, and WATS trunk groups when the Country field is **5**, **16**, or **23**.

Valid entries	Usage
If the Country field is **5**, the Version field only controls TN2147 ports.	
a	Enter **a** to use standard signaling for the Netherlands public network.
b	Enter **b** to use country 1 (U.S.) signaling. The value **b** is appropriate if your switch is connected to a central office using an Ericcson AXE-10 switch.
If the Country field is **16** or **23**, the Version field sets the input impedance value and only controls TN465C (vintage 2 or later) ports.	
a	Enter **a** to set input impedance to 600 Ohms.
b	Enter **b** to set input impedance to 900 Ohms. The value **b** is appropriate in Brazil.

Incoming Destination

Use this field to set the destination for all incoming calls on trunk groups such as CO, FX, and WATS that must terminate at a single destination. The destination you enter here is also the default night service destination unless you enter a different destination in the Night Service field. Appears when the Direction field is **incoming** or **two-way**.

Valid entries	Usage
blank	Leave this field blank if the Trunk Type (in/out) field is not **auto/...**.
an extension number	Calls go to the extension you enter. You may enter any type of extension, though typically the extension entered here identifies a VDN, a voice response unit, or a voice messaging system. Night service overrides this setting when it’s active.
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.

Comm Type

Use this field to define whether the trunk group carries voice, data, or both.

NOTE:
Comm Types of **avd**, **rbavd** and **data** require trunk member ports on a DS1 circuit pack.

Valid entries	Usage
avd	Enter **avd** for applications that mix voice and Digital Communication Protocol data, such as video conferencing applications. The receiving switch discriminates voice calls from data calls and directs each to an appropriate endpoint. Neither originating nor terminating switches insert a modem poll for any calls when Comm Type is **avd**. The Signaling Mode field on the DS1 circuit pack screen must be set for either **common-chan** or **CAS** signaling.
data	Enter **data** only when all calls across the trunk group originate and terminate at Avaya MultiVantage digital data endpoints. Public networks don't support **data**: supported by Avaya's DCP protocol, this entry is used almost exclusively for the data trunk group supporting DCS signaling channels. The Signaling Mode field on the DS1 circuit pack may be set to **robbed-bit** or com**mon-chan**.
rbavd	For digital trunk groups that carry voice and data with robbed-bit signaling. The Signaling Mode field on the DS1 circuit pack screen must be set to **robbed-bit** unless mixed mode signaling is allowed on the DS1 circuit pack. In that case, the Signaling Mode field may be **isdn-ext** or **isdn-pri**.
voice	For trunk groups that carry only voice traffic and voice-grade data (that is, data transmitted by modem). Analog trunk groups must use **voice**.

Auth Code

This field affects the level of security for tandemed outgoing calls at your switch. This field appears if the Direction field is incoming or two-way, and it can only be **y** if the Authorization Codes field is **y** on the System-Parameters Customer-Options screen.

Valid entries	Usage
y/n	Enter **y** to require callers to enter an authorization code in order to tandem a call through an AAR or ARS route pattern. The code will be required even if the facility restriction level of the incoming trunk group is normally sufficient to send the call out over the route pattern.

Digit Absorption List

This field assigns a digit absorption list, when used, to a trunk group that terminates at a step-by-step central office.

Valid entries	Usage
0 to **4**	Enter the number of the digit absorption list this trunk group should use.

Prefix-1

Use this field for outgoing and two-way trunk groups handling long distance service. This field appears for CO, FX, and DIOD trunk groups.

Valid entries	Usage
y/n	Enter **y** to add the prefix “1” to the beginning of the digit string for outgoing calls. Do not enter **y** for trunk groups in AAR or ARS route patterns.

Trunk Flash

Trunk Flash enables multifunction telephones to access central office customized services that are provided by the far-end or Central Office (CO) located directly behind the MultiVantage System. These central office customized services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from the MultiVantage system telephone on an active trunk call.

Valid entries	Usage
y/n	Enter **y** to allow trunk flash.

Toll Restricted

Valid entries	Usage
y	Enter **y** to prevent toll-restricted users from using a trunk access code to make restricted outgoing calls over this trunk group.
n	Enter **n** if the field is automatic or if you don’t want to restrict access.

BCC

Generalized Route Selection uses the BCC to select the appropriate facilities for routing voice and data calls. Far-end tandem switches also use the BCC to select outgoing routing facilities with equivalent BCC classes.The entry in the Bearer Capability Class field is used to select the appropriate facilities for incoming ISDN calls. Avaya MultiVantage compares the entry in the BCC field to the value of the Bearer Capability information element for the incoming call and routes the call over appropriate facilities. For example, a call with BCC 4 will only be connected through facilities that support 64 kbps data transmission.

The Bearer Capability Class field appears when all of the following are true:

- Either the ISDN-BRI Trunks field or the ISDN-PRI field on the System Parameters Customer-Options screen is **y**.
- The Group Type field is **access**, **co**, **fx**, **tandem**, **tie**, or **wats**.
- The Comm Type field is **data**, **avd**, or **rbavd**.

Valid entries	Usage
0	For voice and voice-grade data
1	For 56 kbps synchronous data transmitted with robbed-bit signaling
2	Less than 19.2 kbps synchronous or asynchronous data
3	For 64 kbps data and LAPD protocol
4	For 64 kbps data on unrestricted channels

ITC

The Generalized Route Selection feature, part of the automatic routing technology used in Avaya MultiVantage, compares the line coding of available digital facilities and selects appropriate routes for voice and data calls. The Information Transfer Capability field appears when the Comm Type field is **data**, **avd**, or **rbavd** and the BCC field is not **0**.

Valid entries	Usage
rest(ricted)	Restricted trunks use **ami-basic** or **ami-zcs** line coding and can carry only restricted calls.
unre(stricted)	Unrestricted trunks use **b8zs**, **hdb3,** or **cmi** line coding and can carry restricted or unrestricted calls. A trunk group with an unrestricted ITC may have only unrestricted trunks as members.

Tip:
To find out what kind of line coding a trunk group member uses, check the Line Coding field on the DS1 Circuit Pack screen for the DS1 port to which the member is assigned.

Trunk Type

Use this field to control the seizure and start-dial signaling used on this trunk group. Entries in this field vary according to the function of the trunk group and *must* match the corresponding setting on the far-end switch. This field appears for CO, DID, FX, and WATS trunk groups.

Refer to ''Transmission and supervisory signaling'' on page 1787 for more information. Procedures in ''Managing trunks'' on page 403 give specific suggestions for signaling to use with different types of trunk groups.

Valid entries	Usage
ground-start	Use ground-start signaling for two-way trunks whenever possible: ground-start signaling avoids glare and provides answer supervision from the far end.
loop-start	In general, try to use loop-start signaling only for one-way trunks. Loop-start signaling is susceptible to glare and does not provide answer supervision.
auto/auto **auto/delay** **auto/immed** **auto/wink**	The term before the slash tells the switch how and when it will receive incoming digits. The term after the slash tells the switch how and when it should send outgoing digits. ■ auto — Used for immediate connection to a single preset destination (incoming CO trunks, for example). No digits are sent, because all calls terminate at the same place. ■ delay — The sending switch does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end switch, indicating that it's ready to receive the digits. ■ wink — The sending switch does not send digits until it receives a a wink start (momentary off-hook) signal from the far-end switch, indicating that it's ready to receive the digits. ■ immed — The sending switch sends digits without waiting for a signal from the far-end switch.
2-wire-ac **2-wire-dc** **3-wire**	These entries are used with CO trunks in Russia: enter the type of connection to your central office. Check with you network service provider if you don't know what type of connection they're using. To use these entries, the Country field must be **15** and the CO trunks must use ports on a TN2199 circuit board.

Trunk Type (in/out)

Use this field to control the seizure and start-dial signaling used on this trunk group. The setting of the Trunk Signaling Type field can affect the entries allowed in this field. In addition, settings may differ for incoming and outgoing trunks.

Valid entries	Usage
auto cont **delay** disc **immed** **wink**	There are numerous valid entries for this field: use the online help in the switch administration software to view all the possible combinations. Here are what the elements used in those combinations mean: ■ auto — Used for immediate connection to a single preset destination (incoming CO trunks, for example). No digits are sent, because all calls terminate at the same place. ■ cont — Continuous signaling is used with Italian E&M tie trunks. The switch seizes a trunk by sending a continuous seizure signal for at least the duration specified by the Incoming Seizure Timer. Refer to ''Trunk Signaling Type'' on page 1173 for more information. ■ delay — The sending switch does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end switch, indicating that it's ready to receive the digits. ■ disc —Discontinuous signaling is used with Italian tie trunks that use E&M signaling. The switch can seize a trunk by sending a single, short signal for the duration specified by the Normal Outgoing Seize Send field. However, with the Three-Way Seizure option the calling switch can also send routing information to the called switch by sending one or a series of brief seizure signals. ■ wink — The sending switch does not send digits until it receives a a wink start (momentary off-hook) signal from the far-end switch, indicating that it's ready to receive the digits. ■ immed — The sending switch sends digits without waiting for a signal from the far-end switch.
2-wire-ac **2-wire-dc** **3-wire**	These entries are used with CO trunks in Russia: enter the type of connection to your central office. Check with you network service provider if you don't know what type of connection they're using. To use these entries, the Country field must be **15** and the CO trunks must use ports on a TN2199 circuit board.

Tip:
When incoming trunks use the setting immed/immed, the far-end switch seizes the trunk and sends digits without waiting for acknowledgment from the receiving switch. When traffic is heavy, the receiving switch may not immediately attach a Touch Tone Receiver to a call and therefore lose digits. Use wink-start trunks or increase the dial-guard timer value on the far-end switch to avoid this problem.

Incoming Rotary Timeout (sec)

Call setup at central offices that still use older switching equipment, such as step-by-step technology, is considerably longer then at central offices with more modern switches. If you're receiving digits with incoming calls from a central office that uses less efficient switching technology, your switch needs to allow more time to ensure it receives all the incoming digits. When the Incoming Dial Type field is rotary, use this field to set the maximum time your switch will wait to receive all incoming digits from the far-end switch.

Valid entries	Usage
5 to **99**	If the system is connected to a step-by-step central office, or any CO using older switching technology, enter at least **18** seconds; if not, enter at least **5** seconds.

Outgoing Dial Type

This field sets the method used to transmit digits for an outgoing call. Usually, you should match what your central office provides. Refer to ''Types of address transmission'' on page 1790 for more information. This field appears for Access, APLT, CO, DIOD, DMI-BOS, FX, RLT, and WATS trunk groups. It also appears for Tie trunk groups when the Trunk Signaling Type field is blank, **cont**, or **dis**.

DIOD trunks support pulsed and continuous E&M signaling in Brazil and discontinuous E&M signaling in Hungary.

Valid entries	Usage
tone	Enter **tone** to use Dual Tone Multifrequency (DTMF) addressing, also known as "touch tone" in the U.S. Entering **tone** actually allows the trunk group to support both DTMF and rotary signals. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone** or **mf**.
rotary	Enter **rotary** if you only want to allow the dial pulse addressing method used by non-touch tone phones. If you have a full touch tone system internally and a connection to a central office that only supports rotary dialing, for example, it would be appropriate to enter **rotary**.
r1mf	Enter **r1mf** for CAMA trunk groups. Enter **r1mf** to allow Russian MF Packet Signaling on outgoing trunks. Russian MF Packet Signaling carries calling party number and dialed number information. Group type must be set to co.

Valid entries	Usage
mf	Enter **mf** if the Trunk Signaling Type field is blank. The Multifrequency Signaling field must be enabled on the System-Parameters Customer-Options screen in order for you to enter **mf** here. You may not enter mf if the Used for DCS field (Field descriptions for page 2) is **y**. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone** or **mf**.
automatic	Enter **automatic** for tie trunks if the Trunk Signaling Type field is blank. This provides “cut-through” operation to outgoing callers who dial a trunk access code, connecting them directly to central office dial tone and bypassing any toll restrictions administered on your switch.
r1mf	CAMA trunk groups use **r1mf**. It is the only outgoing dial type allowed on CAMA trunk groups.

Cut-Through

This field appears when the Outgoing Dial Type field is either **rotary** or **tone**.

SECURITY ALERT:

*Entering **y** in this field will reduce your ability to prevent toll fraud.*

Valid entries	Usage
y	Enter **y** to allow users to get dial tone directly from the central office. Outgoing calls over this trunk group will bypass AAR/ARS (if you’re using it) and any of your administered restrictions (such as COR or FRL).
n	Enter **n** and the user will receive switch dial tone. Instead of digits being sent to the central office, they will be collected and checked against administered restrictions. If no restrictions apply, the digits are sent to the central office.

Incoming Dial Type

Indicates the type of pulses required on an incoming trunk group. Usually, you should match what your central office provides. Refer to ''Types of address transmission'' on page 1790 for more information. This field appears for Access, APLT, DID, DIOD, DMI-BOS, FX, RLT, Tandem, and WATS trunk groups. It also appears for Tie trunk groups when the Trunk Signaling Type field is blank, **cont**, or **dis**.

Valid entries	Usage
tone	Enter **tone** to use Dual Tone Multifrequency (DTMF) addressing, also known as "touch tone" in the U.S. Entering **tone** actually allows the trunk group to support both DTMF and rotary signals. Also, if you're using the Inband ANI feature, enter **tone**. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone**.
rotary	Enter **rotary** if you only want to allow the dial pulse addressing method used by non-touch tone phones. Though the tone entry supports rotary dialing as well, it's inefficient to reserve touch tone registers for calls that don't use DTMF.
mf	Enter **mf** if the Trunk Signaling Type field is blank. The Multifrequency Signaling field must be enabled on the System-Parameters Customer-Options screen in order for you to enter **mf** here. You may not enter **mf** if the Used for DCS field (Field descriptions for page 2) is **y**. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **mf**.

NOTE:
The value in this field affects the appearance of the Incoming Partial Dial (sec) field on the Administrable Timer Page.

Wink Timer (msec)

This field allows you to reduce the risk of glare by controlling part of call setup. Requirements for the United States domestic network specify that the wink signal for wink-start trunks must begin within 5 seconds after a trunk is seized. For trunks with a delay-dial start, the wink must not last longer than 5 seconds. While some circuit packs are hard-coded to allow the full 5 seconds in both cases, other circuit packs allow you reduce the allowed start time and duration, thus reducing the window in which glare could occur.

Unlike other fields on this screen, the Wink Timer field therefore controls 2 different variables. What your entry does depends on the outgoing value in the Trunk Type (in/out) field.

Setting of the Trunk Type (in/out) field	What the Wink Timer field sets
…/wink	Maximum duration of the wink signal (wait-for-wink-to-end)
…/delay	Maximum interval after trunk seizure for the wink to begin (wait-for-wink-to-start)

This field appears when the “out” side of the entry in the Trunk Type (in/out) field is **…/wink** or **…/delay** and the Group Type is **tie**, **access**, **aplt**, **dmi-bos**, **rlt**, or **tandem**. The setting in this field only affects trunks administered to ports on TN760C (vintage 4 or later), TN767, TN464C (or later), and TN2242 circuit packs. If the trunk group also contains trunks assigned to ports on other types of circuit packs, those trunks are unaffected.

Valid entries	Usage
300 to **5000** in increments of 50	In general, Avaya recommends that you not change this field. If you do, remember that your switch’s timing must be compatible with the timing on the far-end switch.

Trunk Termination

This field adjusts the impedance of the trunk group for optimal transmission quality. Check with your service provider if you’re not sure of the distance to your central office.

Valid entries	Usage
600ohm	Enter **600ohm** when the distance to the central office or the switch at the other end of the trunk is less than 3,000 feet.
rc	Enter **rc** when the distance to the central office or the switch at the other end of the trunk is more than 3,000 feet.

Disconnect Timing (msec)

This field specifies the minimum time in milliseconds that the central office or far-end switch requires to recognize that your switch has disconnected from a call. This timer does not affect ports on a circuit pack that uses the administrable Incoming Disconnect and Outgoing Disconnect timers; in fact, settings on those two timers override this field.

Valid entries	Usage
140 to **2550** ms in increments of 10	The default of **500** is an industry standard and you shouldn't change it. If you set this field too high, your switch won't disconnect sometimes when it should; too low, and it will disconnect when it shouldn't.

End-to-End Signaling

Auxiliary equipment such as paging equipment and music sources may be connected to Avaya MultiVantage by auxiliary trunks. The switch may send DTMF signals (touch tones) to these devices. This field, which appears for CPE (customer-provided equipment) trunk groups, sets the duration of these tones.

Valid entries	Usage
60 to **360** ms in increments of 10	Use this field to set the duration in milliseconds of the touch-tone signal that is sent to the connected equipment.

NOTE:
For trunks that do not receive real answer supervision, a "connect" Event report is sent when the Answer Supervision Timeout occurs.

Analog Gain

Use this field to reduce the strength of incoming signals on TN2199 ports if users regularly experience echo, distortion, or unpleasantly loud volume. Experiment to find the best setting for your situation. This field appears if the Country field is **15** and the Trunk Type (in/out) field is **2-wire-ac**, **2-wire-dc**, or **3-wire**.

Valid entries	Usage
a	Reduces the incoming signal by -3dB.
b	Reduces the incoming signal by -6dB.
c	Reduces the incoming signal by -8dB.
none	No reduction. Don't change this setting unless the trunk group's sound quality is unacceptable.

Digit Treatment

Use this field to modify an incoming digit string (as on DID and tie trunks, for example) by adding or deleting digits. You'll need to do this if the number of digits you receive doesn't match your dial plan.

Valid entries	Usage
blank	The incoming digit string is not changed.
absorption	Deletes digits, starting at the beginning of the string.
insertion	Adds digits, starting at the beginning of the string.

If you enter absorption or insertion, then you must enter a value in the Digits field.

NOTE:
In a DCS network, DCS features that use the remote-tgs button (on phones at a remote switch) do not work when the incoming trunk group at your switch deletes or inserts digits on incoming calls. The remote-tgs button on a remote switch, for example, tries to dial a TAC on your switch. If your switch adds or deletes digits, it defeats this operation. If you need to manipulate digits in a DCS network (for example, to insert an AAR feature access code), do it on the outgoing side based on the routing pattern.

Digits

This field is used with the Digit Treatment field, and its meaning depends on the entry in that field. If the Digit Treatment field is **absorption**, this field specifies *how many* digits are deleted. If the Digit Treatment field is **insertion**, this field identifies the *specific digits* that are added.

Valid entries	Usage
1 to **5**	Enter the number of digits to be deleted (absorbed).
Up to 4 digits, including * and #	Enter the actual digits to be added (inserted).
blank	This field can be blank only if the Digit Treatment field is blank.

Expected Digits

> **NOTE:**
> Set this field to **blank** if the Digit Treatment field is set to **insert** and the Digits field contains a feature access code (for example, AAR or ARS) followed by digits. In this case, the number of digits expected are set on the AAR and ARS Digit Analysis Table and AAR and ARS Digit Conversion Table.

Valid entries	Usage
1 to **18**	Enter the number of digits that the far-end switch sends for an incoming connection. If your switch is absorbing digits on this trunk group, the entry in this field must be larger than the entry in the Digits field.
blank	If you leave this field blank, you cannot administer digit absorption.

Auto Guard

This field controls ports only on TN438B, TN465B, and TN2147 circuit packs. TN438B ports have hardware support for detecting a defective trunk. TN465B and TN2147 ports consider a trunk defective if no dial tone is detected on an outgoing call, and the Outpulse Without Tone field is **n** on the Feature-Related System Parameters screen.

Valid entries	Usage
y/n	Enter **y** to prevent repeated seizures of a defective trunk. The switch will do a maintenance busy-out on these trunks.

Call Still Held

This field is used when the receiving switch initiates the disconnection of incoming calls. It effectively extends the Incoming Glare Guard timer by 140 seconds. This field affects only TN438B, TN465B, and TN2147 ports and is used primarily when the Country Code field is **2**.

Valid entries	Usage
y/n	Enter **y** to prevent glare by delaying an outgoing seizure of a trunk for at least 140 seconds after it is released from an incoming call.

Sig Bit Inversion

When transmission facilities use bit-oriented signaling (such as CAS), 2 bits are used to transmit seizure and release signals for calls. Called the A-bit and the B-bit, their meaning can vary. For example, in the A-bit a "1" might mean on-hook and a "0" might mean off-hook. The entry in the Country Protocol field on the DS1 Circuit Pack screen sets the default meaning of these bits.

For trunk ports on TN2242 and TN464B and later circuit packs, this field allows you to invert the A- and B-bits as necessary so that the far-end switch can understand your switch's seizure and release signals. If the far-end switch, such as a central office, on this trunk group interprets the A- and B-bits differently from the default, you may need to invert one or both bits — to change "1" to "0" and vice-versa in the A-bit, for example.

Valid entries	Usage
A **B** **A&B** **none**	For the TN464B and later circuit packs, indicate which bits, if any, should be inverted.
A and **none**	For the Japanese 2Mbit trunk circuit pack, indicate which bits, if any, should be inverted.

Analog Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over an analog signaling port in the trunk group.

Valid entries	Usage
1 to **17**	Shows the index into the loss plan and tone plan.

Digital Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over a digital signaling port in the trunk group.

Valid entries	Usage
1 to **17**	Shows the index into the loss plan and tone plan.

Incoming Dial Tone

Indicates whether or not your switch will give dial tone in response to far-end seizures of the trunk group.

Valid entries	Usage
y	Enter **y** if the incoming trunk group transmits digits. For example, you would enter **y** for two-way, dial-repeating tie trunks that users select by dialing a trunk access code.
n	Enter **n** for trunks that aren't sending digits, such as tandem or incoming CO trunks.

Line Length

This field appears only when the Group Type field is **tie** and the Trunk Signaling Type field is **tge**, **tgi**, or **tgu**.

Valid entries	Usage
short	
long	

NOTE:
Unless one or more trunk members are administered, the administered value is not saved when you submit the screen (press ENTER).

Send Release Ack

This field appears when the Trunk Signaling Type field is **cont** or **dis** and only applies to TN2140 ports (used for Italian and Hungarian tie trunks).

Valid entries	Usage
y/n	Enter **y** to send a release acknowledgment in response to a forward or backward release signal.

Receive Release Ack

This field appears when the Trunk Signaling Type field is **cont** or **dis** and only applies to TN2140 ports (used for Italian and Hungarian tie trunks).

Valid entries	Usage
y/n	Enter **y** if the switch will receive a release acknowledgment in response to a forward or backward release signal.

Dial Detection

Applies only to TN2199 ports. The Country field must be **15**.

Valid entries	Usage
A-wire	
B-wire	

Extended Loop Range

This field appears only for a DID trunk group and is used only with the TN459A circuit pack. Enter **y** or **n** depending on the distance between the central office and the switch. If greater than the required distance, then the field should be **y**.

Valid entries	Usage
y/n	

Trunk Gain

This field specifies the amplification applied to the trunks in this group. With the values of the Trunk Termination and Country fields, the value in this field also determines the input and trans-hybrid balance impedance for TN465B, TN2146, TN2147, and TN2184 ports. All other CO and DID circuit packs are set automatically to high

Valid entries	Usage
high	Enter **high** if users complain of low volume.
low	Enter **low** if users complain of squeal or feedback.

Drop Treatment

This field only applies to DID trunks. It determines what the calling party hears when the called party terminates an incoming call.

Valid entries	Usage
intercept **busy** **silence**	Select one. For security reasons, it's better to apply a tone: silence could provide an opening for hackers.

NOTE:
In Italy, the Drop Treatment field must be administered as **intercept** for all DID trunk groups.

Bit Rate

This field specifies the baud rate to be used by pooled modems. This field appears when the Comm Type field is **avd** or **rbavd**. It also appears if the Comm Type field is **data**, but only if the ISDN-PRI field is **y** on the System-Parameters Customer-Options screen.

Valid entries	Usage
300 **1200** **2400** **4800** **9600** **19200**	Enter the speed of the fastest modem that will use this trunk group.

Synchronization

This field determines whether the trunk group will use synchronous or asynchronous communications. This field appears if:

- the Group Type field is **dmi-bos** or **isdn**
- the Group Type field is **access**, **co**, **fx**, **tandem**, **tie**, or **wats** and the Comm Type field is **avd** or **rbavd**.
- the Group Type field is **access**, **co**, **fx**, **tandem**, **tie**, or **wats**, the Comm Type field is **data**, and the ISDN-PRI field or the ISDN-BRI Trunks field is **y** on the System-Parameters Customer-Options screen.

Valid entries	Usage
async **sync**	Do not change this field without the assistance of Avaya or your network service provider.

Duplex

This field specifies whether a two-way trunk group allows simultaneous transmission in both directions. This field appears when the Comm Type field is **avd** or **rbavd**. It also appears if the Comm Type field is **data**, but only if the ISDN-PRI field is enabled on the System-Parameters Customer-Options screen.

NOTE:
Even if the trunk group supports full-duplex transmission, other equipment in a circuit may not.

Valid entries	Usage
full	Enter **full** in most cases: this allows simultaneous two-way transmission, which is most efficient.
half	Enter **half** to support only one transmission direction at a time.

Disconnect Supervision-In

This field indicates whether the switch receives disconnect supervision for incoming calls over this trunk group. It appears when the Direction field is **incoming** or **two-way**. (If the Direction field is **outgoing**, the switch internally sets this field to **n**.)

The entry in this field is crucial if you allow trunk-to-trunk transfers. (To allow trunk-to-trunk transfers involving trunks in this group, this field must be **y** and the Trunk-to-Trunk Transfer field on the Feature-Related System Parameters screen must be **y**). If a user connects 2 trunks through conference or transfer, and neither far-end switch on the resulting connection provides disconnect supervision, the trunks involved will not be released because Avaya MultiVantage can't detect the end of the call. Avaya MultiVantage will not allow trunk-to-trunk transfers unless it believes that at least one party on the call can provide disconnect supervision. Therefore, setting this field incorrectly may cause trunks to become unusable until the problem is detected and the trunks are reset.

Valid entries	Usage
y	Enter **y** to allow trunk-to-trunk transfers involving trunks in this group. Enter **y** if the far-end switch sends a release signal when the calling party releases an incoming call, and you want to make the far-end switch responsible for releasing the trunk. Enter **y** to enhance Network Call Redirection.
n	Enter **n** if the far-end switch doesn't provide a release signal, if your hardware can't recognize a release signal, or if you prefer to use timers for disconnect supervision on incoming calls. Entering **n** prevents trunk-to-trunk transfers involving trunks in this group.

CAUTION:

In general, U.S. central offices provide disconnect supervision for incoming calls but not for outgoing calls. Public networks in most other countries do not provide disconnect supervision for incoming or outgoing calls. Check with your network services provider.

Disconnect Supervision-Out

This field indicates whether the switch receives disconnect supervision for outgoing calls over this trunk group. It appears when the Direction field is either **outgoing** or **two-way**. (If the Direction field is **incoming**, the switch internally sets this field to **n**.)

The entry in this field is crucial if you allow trunk-to-trunk transfers. (To allow trunk-to-trunk transfers involving trunks in this group, this field must be **y** and the Trunk-to-Trunk Transfer field on the Feature-Related System Parameters screen must be **y**). If a user connects 2 trunks through conference or transfer, and neither far-end switch on the resulting connection provides disconnect supervision, the trunks involved will not be released because Avaya MultiVantage can't detect the end of the call. Avaya MultiVantage will not allow trunk-to-trunk transfers unless it believes that at least one party on the call can provide disconnect supervision. Therefore, setting this field incorrectly may cause trunks to become unusable until the problem is detected and the trunks are reset.

Also, remember that Avaya MultiVantage must receive *answer* supervision on outgoing analog CO, FX, WATS, Tie, Tandem, and Access trunks before it will recognize a disconnect signal. If this trunk group does not receive *answer* supervision from the far-end switch, and you enter **y** in this field, Avaya MultiVantage will internally set the field to **n**.

Valid entries	Usage
y	Enter **y** to allow trunk-to-trunk transfers involving trunks in this group. Enter **y** if the far-end switch sends a release signal when the called party releases a call an outgoing call, and you want to make the far-end switch responsible for releasing the trunk. The Answer Supervision Timeout field must be **0** and the Receive Answer Supervision field must be **y** for the switch to recognize a **y** entry. Enter **y** to enhance Network Call Redirection.
n	Enter **n** if the far-end switch doesn't provide a release signal, if your hardware can't recognize a release signal, or if you prefer to use timers for disconnect supervision on outgoing calls. Entering **n** prevents trunk-to-trunk transfers involving trunks in this group.

CAUTION:

Do not set this field to **y** *unless you are certain that the far-end switch will provide answer supervision and disconnect supervision. Most public networks do not provide disconnect supervision over analog trunks. Check with your network services provider.*

Cyclical Hunt

When a call is offered to a trunk group, the switch searches for an available trunk. This field, which appears when the Direction field is **two-way** and the Trunk Type field is **loop-start**, controls the starting point of the search.

You can change this field from **n** to **y** at any time. To change from **y** to **n**, however, all the trunks in the group must be idle or busied out.

Valid entries	Usage
y	Enter **y** to have the switch start its search from the last trunk seized. This method is faster, and thus better suited for high-traffic trunk groups.
n	Enter **n** to have the switch start each search at member 1 (the first trunk administered on the Group Member Assignments page).

Disconnect Type

This field indicates which side or user controls the disconnect, where A refers to the *calling* party and B refers to the *called* party. Appears only if the Country field is **15** and the Trunk Type field is **2-wire-ac**, **2-wire-dc**, or **3-wire**.

This applies *only* to the TN2199 port.

Valid entries	Usage
A and B	
A or B	

Send Answer Supervision

This field appears when the Trunk Signaling Type field is **cont** or **dis** and only applies to TN2140 ports.

Valid entries	Usage
y/n	Enter **y** to make your switch signal the calling switch when an incoming call is answered. You can only set this field to **y** if the Direction field is **incoming** or **two-way**.

Answer Supervision Timeout

If the Receive Answer Supervision field is **n**, use this field to set the answer supervision timer for outgoing and two-way trunks. During a cut-through operation, timing begins after each outgoing digit is sent by the switch and timing ceases after the far-end sends answer supervision. If the timer expires, the switch acts as if it had received answer supervision. On senderized operation, the timer begins after the last digit collected is sent by the switch.

Valid entries	Usage
0 to **250**	Enter the number of seconds you want the switch to wait before it acts as though answer supervision has been received from the far-end. Set this field to **0** if Receive Answer Supervision is **y**.

NOTE:
This field's setting does not override answer supervision sent from the network or from DS1 port circuit timers. To control answer supervision sent by DS1 firmware, set the Outgoing End of Dial (sec) field on the Administrable Timers page of the trunk group screen.

Receive Answer Supervision

Use this field to specify whether the network provides answer supervision for a trunk group.

For Outbound Call Management applications, set this field to **y** for trunks supporting network answer supervision. For trunks that do not receive a real answer, this field determines when the CallVisor Adjunct-Switch Application Interface (ASAI) connect event is sent.

Valid entries	Usage
y	Enter **y** if the network provides answer supervision. Set the Answer Supervision Timeout field to **0.**
n	Enter **n** if the network does not provide answer supervision, and set the Answer Supervision Timeout field. Also enter **n** for incoming trunk groups.

NOTE:
When you set this field to **y**, the Outgoing End of Dial (sec) field is not displayed. The firmware timeout on circuit packs controlled by the Outgoing End of Dial (sec) field is automatically set to **0**.

Supplementary Service Protocol

Appears only when trunk group Type is ISDN.

Valid entries	Usage
a	Allows ASAI Flexible Billing.
	AT&T, Bellcore, Nortel.
	When the Country Code field on the DS1 screen is 1A, SSA selects AT&T custom supplementary services.
	When the Country Code field on the DS1 screen is 1B, SSA selects Bellcore Supplementary Services.
	When the Country Code field on the DS1 screen is 1C, SSA selects Nortel Proprietary Supplementary Services.
b	ISO QSIT
c	ETSI
	Use c protocol for Network Call Deflection. For more information, see *Avaya MultiVantage Network Call Redirection* (555-233-759).
d	ECMA QSIG
e	Allows ASAI Flexible Billing.
	Allows DCS with rerouting. DCS with Rerouting must be y, and the Used for DCS field on the trunk group screen must be y.
f	Feature Plus
g	ANSI. Available only if, on the System-Parameters Customer-Options screen, the ISDN-PRI or ISDN-BRI field is **y** or the Used for DCS field is **y**.
	Use g protocol for Network Call Transfer. For more information, see *Avaya MultiVantage Network Call Redirection* (555-233-759).

Field descriptions for page 2

The figure below shows a common configuration for page 2 of the Trunk Group screen when the Group Type field is **co**. This screen is only an example, and the fields shown below may change or disappear according to specific field settings.

```
add trunk-group next                                                   Page 2 of x
TRUNK FEATURES
           ACA Assignment? _         Measured: ____     Wideband Support? _
 Long Holding Time(hours): _                           Maintenance Tests? _
  Short Holding Time(sec): _  Data Restriction? _    NCA-TSC Trunk Member: _
  Short Holding Threshold: __        Send Name: _     Send Calling Number: _
             Used for DCS? _
    Suppress # Outpulsing? _  Numbering Format: _______
Outgoing Channel ID Encoding: _________     UUI IE Treatment: ______
                                         Maximum Size of UUI IE Contents: ___
                                               Replace Restricted Numbers? _
                                              Replace Unavailable Numbers? _
                                                    Send Connected Number: _

              Send UUI IE? _
               Send UCID? _              BRS Reply-best DISC Cause Value: __
  Send Codeset 6/7 LAI IE? _                        Ds1 Echo Cancellation? _

                                        US NI Delayed Calling Name Update? _

                          Network (Japan) Needs Connect Before Disconnect? _
```

Screen 266. CO Trunk Group

CAUTION:
Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

ACA Assignment

Valid entries	Usage
y/n	Enter **y** if you want Automatic Circuit Assurance (ACA) measurements to be taken for this trunk group. If **y** is entered, complete the Long Holding Time, Short Holding Time, and Short Holding Threshold fields.

Measured

Indicates if the system will transmit data for this trunk group to the Call Management System (CMS). You cannot use **internal** and **both** unless either the BCMS (Basic) or the VuStats field is **y** on the System-Parameters Customer-Options screen. If the ATM field is set to **y** on the System-Parameters Customer-Option screen, this field accepts only **internal** or **none**. If this field contains a value other than **internal** or **none** when ATM is **y**, **none** appears.

Valid entries	Usage
internal	Enter **internal** if the data can be sent to the Basic Call Management System (BCMS), the VuStats data display, or both.
external	Enter **external** to send the data to the CMS.
both	Enter **both** to collect data internally and to send it to the CMS.
none	Enter **none** if trunk group measurement reports are not required.

Long Holding Time (hours)

This field appears only when the ACA Assignment field is **y**.

Valid entries	Usage
0 to **10**	Enter the length of time (in hours) that the system will consider as being a long holding time. If you enter **0**, the system will not consider long holding calls.

Internal Alert

Valid entries	Usage
y/n	Enter **y** if internal ringing and coverage will be used for incoming calls.

Maintenance Tests

Valid entries	Usage
y/n	Enter **y** if hourly maintenance tests will be made on this trunk group. Your entry is not saved when the screen is submitted unless one or more trunk members are administered.

Short Holding Time (seconds)

This field appears when the ACA Assignment field is **y**.

Valid entries	Usage
0 to **160**	Enter the length of time (in seconds) that the system considers as being a short holding time. If **0** is entered, the system will not consider short holding calls.

Data Restriction

If **y**, whisper page is denied on this trunk.

Valid entries	Usage
y/n	Enter **y** to prevent features from generating tones on a data call that would cause erroneous data transmission.

Short Holding Threshold

This field appears when the ACA Assignment field is **y**.

Valid entries	Usage
0 to **30**	Enter the number of times the system will record a short holding call before alerting an attendant to the possibility of a faulty trunk.

Glare Handling

This field determines what the switch will do when glare occurs. This field appears when the Direction field is **two-way** and the outgoing side of the Trunk Type field is either **.../wink** or **.../delay**.

If you enter **control** or **backoff**, and ports for the trunk group are not capable of detecting glare, warnings are generated. The following circuit packs can detect glare: TN767 (all releases), TN760C (or later releases), and TN464C (or later releases).

Valid entries	Usage
control	Your switch will seize the trunk and proceed with call setup. The other switch will find another trunk.
backoff	The other switch will seize the trunk proceed with call setup. Your switch will find another trunk.
none	

Abandoned Call Search

Use this field when the Trunk Type field is **ground-start**. Abandoned Call Search is designed to work with analog ground-start CO trunks that *do not* provide disconnect supervision. Your central office must support Abandoned Call Search for the feature to work properly. If your central office provides disconnect supervision, you do not need to use the Abandoned Call Search feature.

Valid entries	Usage
y/n	Enter **y** if this trunk group will conduct an Abandoned Call Search to identify ghost calls.

Used for DCS

Valid entries	Usage
y/n	Enter **y** if this trunk group will send and receive messages on a DCS signaling link.

NOTE:
This field cannot be activated if the trunk group number is greater than 255 or if the Trunk Access code is more than 3-digits long.

If this field is **y**, you can administer ISDN-BRI trunk groups unless the DCS Signaling field is **d-chan**. In that case, remove the BRI trunks or set the DCS Signaling field to **bx.25** before submitting the screen.

PBX ID

This field, which appears when the Used for DCS field is **y**, identifies the remote switch in the network with which the trunk will communicate on a DCS signaling link.

Valid entries	Usage
1 to **63**	Enter the ID of the switch at the other end of this trunk.

Suppress # Outpulsing

Valid entries	Usage
y/n	Enter **y** if end-to-end signaling begins with (and includes) “#”. The final “#” is suppressed in cases where the system would normally outpulse it. This field should be **y** when the Central Office (for example, rotary) or any other facility treats “#” as an error.

Seize When Maintenance Busy

This field only affects ports on TN760C (or later release), TN767, and TN464C (or later release) circuit packs. It indicates whether this switch generates an outgoing seizure when a trunk in this trunk group is maintenance busied and whether the far-end switch is administered to do likewise. This supports the Electronic Tandem Network Busyout feature, which is intended to prevent a far-end switch from reporting problems with a trunk that has been removed from service on your switch. This field's setting does not affect the behavior of the far-end switch; it controls the behavior of this switch and defines the *expected* far-end behavior.

For DIOD trunks using TN464F (or later release) or TN2464, displays only when the Group Type field is **diod** and the Trunk Signaling Type field is **pulsed**, **cont**, or **dis.**

Valid entries	Usage
near-end	Enter **near-end** if this switch generates an outgoing seizure when a trunk is maintenance busied but the far-end switch does not. The seizure is maintained until the maintenance busyout is released.
far-end	Enter **far-end** if the far-end switch generates an outgoing seizure when a trunk is maintenance busied but this switch does not.
both-ends	Enter **both-ends** if both this switch and the far-end switch generate an outgoing seizure when a trunk is maintenance busied.

If a switch generates an outgoing seizure when a trunk is busied out, the seizure will probably cause alarms at the far-end switch, perhaps leading to a far-end maintenance busy out, unless the far-end switch is administered to expect this behavior.

If the administered value of this field is either **far-end** or **both-ends**, any abnormally long incoming seizure (including failure to drop from a completed call) is *assumed to be the result of a far-end maintenance busy condition*. Note that this assumption may be incorrect, since the abnormally long seizure may actually be due to failure of the trunk circuit. Regardless of the cause of the abnormally long seizure, your switch does the following things:

1. Generates a warning alarm indicating that the trunk is assumed to be maintenance busy at the far-end
2. Removes the trunk from service
3. Keeps the trunk out of service until a far-end disconnect is received

Allowable values depend on the entry in the Direction field: check the online help in the switch administration software.

Shuttle

This field appears when the Country field is **15** and the Outgoing Dial Type field is **rotary**. It can be administered on TN464D (or later release) or TN2199 circuit packs.

Valid entries	Usage
y/n	Enter **y** to enable MF shuttle signaling.

Charge Conversion

Avaya MultiVantage multiplies the number of charge units by the value of this field and displays it as a currency amount. If there is no value in this field, Avaya MultiVantage displays the number of charge units without converting it to currency. This field appears for CO, DIOD, FX, and WATS trunk groups when the Direction field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the Charge Advice field is *not* **none**.

Valid entries	Usage
1 to **64**,**500**	Enter the value of a charge unit in terms of the currency you use.

Decimal Point

This field appears for CO, DIOD, FX, and WATS trunk groups when the Direction field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the Charge Advice field is *not* **none**.

Valid entries	Usage
comma **period** **none**	Chose the appropriate representation for a decimal point as it will appear on phone displays. Entering **comma** or **period** in this field divides the charge value by 100.

Start B Signal

This field appears when the Country field is **15** and the Shuttle field is **y**. Enter **1**–**3** to indicate which B-signal should be used to start a call. The value administered in this field must be coordinated with your central office. Refer to ‘‘Start Position’’ on page 1205.

Valid entries	Usage
1	Start calls with signal B1 (first digit)
2	Start calls with signal B2 (next digit)
3	Start calls with signal B3 (previous digit)

Currency Symbol

This field appears for CO, DIOD, FX, and WATS trunk groups when the Direction field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the Charge Advice field is *not* **none**.

Valid entries	Usage
1–3 characters (leading and embedded spaces count as characters)	Enter the symbol you want to appear on phone displays before the charge amount.

Request Category

This field appears when the Country field is **15** and the Shuttle field is **y**.

Valid entries	Usage
y/n	Enter **y** if the switch should request a call category from the central office.

Start Position

The value administered in this field must be coordinated with your central office. This field appears when the Country field is **15** and the Shuttle field is **y**.

Valid entries	Usage
1 to **9**	Indicate which digit in the digit string is considered to be the “previously sent” digit (refer to ‘‘Start B Signal’’ on page 1204).

Charge Type

Entries in this field are text strings you use to describe charges related to a phone call. This field appears for CO, DIOD, FX, and WATS trunk groups when the Direction field is **outgoing** or **two-way**. For ISDN trunk groups, it appears when the Charge Advice field is *not* **none**.

Valid entries	Usage
1–7 characters (embedded spaces count as characters)	Enter the words or characters you want to appear on phone displays after the charge amount. Most likely you will use either the currency symbol or the charge type, bur not both.

Receive Analog Incoming Call ID

Your switch stores and displays 15 characters of name and number information associated with an incoming call on analog trunks (ICLID, or incoming call line identification information). This field appears for CO, DID, and DIOD trunk groups when the Analog Trunk Incoming Call ID field on the System-Parameters Customer-Options screen is **y** and the Direction field is **incoming** or **two-way**.

Valid entries	Usage
Bellcore	Used to collect ICLID information in the U.S.
NTT	Used to collect ICLID information in Japan.
disabled	Stops the collection of ICLID information on analog trunks.

Incoming Tone (DTMF) ANI

This field appears only when the Incoming Dial Type field is **tone**. Digits received through Automatic Number Identification (ANI) are printed on a CDR record, passed to the Intuity AUDIX and ASAI interfaces, and displayed on the phone (and on tandem calls if the outgoing trunk requires ANI). Then the digits are sent to the outgoing trunk.

Valid entries	Usage
ANI*DNIS	If 555-3800 calls extension 81120, the trunk group receives *55538000*81120*. The phone displays Call from 555-3800.
ANI*DNIS*	If 555-3800 calls extension 81120, the trunk group receives 55538000*81120*. The phone displays Call from 555-3800.
no	

Per Call CPN Blocking Code

For Access, APLT, CO, DIOD, FX, tandem, tie, and WATS trunk groups only.

Valid entries	Usage
1 to 4 digit number	
***, #**	May be used as the first digit

Connected to CO

This field appears when the Group Type field is **tie**.

Valid entries	Usage
y/n	Enter **y** to allow overlap sending to a Central Office.

Per Call CPN Unblocking Code

For access, APLT, CO, DIOD, FX, tandem, tie, and WATS trunk groups only.

Valid entries	Usage
1 to 4 digit number	
***, #**	May be used as the first digit

Time (sec) to Drop Call on No Answer

This field appears if the Group Type field is **co** or **diod** and the Outgoing Dial Type field is **mf**, or if the Group Type field is **co**, **diod**, **fx**, **wats**, and the Country field is **15**.

Valid entries	Usage
0–**1200**	Enter the duration (in seconds) the switch should wait for outgoing calls to be answered. If the call is not answered in the specified number of seconds, the call drops. If this field is **0**, the timer is not set and no calls drop.

MF Tariff Free

This field appears for Access, APLT, DID, DIOD, DMI-BOS, and Tandem trunk groups when the Incoming Dial Type field is **mf** or the Group Type field is **tie**, the Trunk Signaling Type field is blank, **cont**, or **dis**, and the Incoming Dial Type field is **mf**.

Valid entries	Usage
y/n	Enter **y** to make the switch generate an MFC Tariff-Free Backward Signal (administered on the Multifrequency-Signaling-Related-System- Parameters screen) during call setup instead of the “free” signal. This aids CO billing.

Outgoing ANI

If this trunk group is used for an outgoing call with ANI, the entry in this field overrides the normal ANI. The ANI is sent exactly as administered, except for the normal truncation to 7 digits for Russian ANI. This ANI override works both for calls originated in Avaya MultiVantage and calls tandemed through it. This field appears for CO, DIOD, FX, and WATS trunk groups.

Valid entries	Usage
1 to 15 digits	Enter the digit string to be sent in place of normal ANI.
blank	Leave this field blank to allow ANI to work normally.

DS1 Echo Cancellation

Reduces voice call echo.

Valid entries	Usage
y/n	Enter **y** to allow echo cancellation on a per port basis.

Path Replacement with Retention

Appears when the following fields are set on the Trunk Group screen: trunk group Type is **ISDN**, Supplementary Service Protocol is **b** or **e**, and the Supplementary Services with Rerouting field or the DCS with Rerouting field on the System Parameters Customer Options screen is **y**.

Valid entries	Usage
y/n	Enter **y** to retain the original trunk group. Set to **n** to allow path replacement according to setting on the Path Replacement Method field.

Path Replacement Method

Appears when the following fields are set on the Trunk Group screen: trunk group Type is **ISDN**, Supplementary Service Protocol is **b** or **e**, the Path Replacement with Retention is **n**, and the Supplementary Services with Rerouting field or the DCS with Rerouting field on the System Parameters Customer Options screen is **y**.

Valid entries	Usage
always	Use any QSIG (SSB) trunk group as the replacement trunk group. A new call is always originated, even when the original trunk group is determined to be the replacement trunk group.
BR (better route)	Route pattern preferences help determine trunk group path replacement. The original trunk group is retained if the Path Replacement with Retention field is **y**. Path replacement fails (and the original trunk group is retained) if the Path Replacement with Retention field is **n**.

Network Call Redirection

For more information, see *Avaya MultiVantage Network Call Redirection* (555-233-759).

Valid entries	Usage
deflect	Use to allow Network Call Deflection.
ANSI-transfer	Use to allow Network Call Transfer for MCI DEX 600 ISDN trunks.
Nortel-transfer	Use to allow Network Call Transfer for MCI DMS 250 switches.

Field descriptions for page 3

The figure below shows a common configuration for page 3 of the Trunk Group screen when the Group Type field is **co**. This screen is only an example, and the fields shown below may change or disappear according to specific field settings.

```
add trunk-group next                                                 Page 3 of x
ADMINISTRABLE TIMERS
                Send Incoming/Outgoing Disconnect Timers to TN465 Ports? _
                                              Outgoing Dial Guard(msec): _____
  Incoming Glare Guard(msec): _____          Outgoing Glare Guard(msec): _____

                                   Outgoing Rotary Dial Interdigit (msec): ____
       Ringing Monitor(msec): _____                Incoming Seizure(msec): ____
   Outgoing End of Dial(sec): ___        Outgoing Seizure Response(sec): ___
 Programmed Dial Pause(msec): _____        Disconnect Signal Error(sec): ___
          Flash Length(msec): ____
                                                      Busy Tone Disconnect?

END TO END SIGNALING
    Tone (msec): ____     Pause (msec): 150

OUTPULSING INFORMATION
     PPS: 10    Make(msec): 40    Break(msec): 60  PPM? y  Frequency: 50/12k
```

Screen 267. Administrable Timers for Trunk Group

⚠ CAUTION:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Send Incoming/Outgoing Disconnect Timers to TN465 Ports

The field appears only for a co, fx, or wats trunk group.

Valid entries	Usage
y/n	Enter **y** if you want to send the incoming disconnect and outgoing disconnect timer values to the trunk group ports that are on a TN465 board.

Incoming Disconnect (msec)

The field appears only for an incoming or two-way trunk group when the Trunk Signaling Type field is either blank or **cont**.

Valid entries	Usage
50 to **2550** in increments of 10	Enter the minimum valid duration of a disconnect signal for an incoming call. The switch will not recognize shorter disconnect signals. This field cannot be blank. For Brazil pulsed E&M signaling, use **600**.

Outgoing Disconnect (msec)

Valid entries	Usage
50 to **2550** in increments of 10	Enter the minimum valid duration of a disconnect signal for an outgoing call. The switch will not recognize shorter disconnect signals. This field cannot be blank. This timer begins timing when a disconnect signal is detected on an outgoing call and resets when the signal is no longer detected. If the timer expires, the trunk drops. For Brazil pulsed E&M signaling, use **600**.

Cama Outgoing Dial Guard (msec)

This field appears when Group Type is **cama** (the trunk group type used for emergency 911 service).

Valid entries	Usage
25 to **6375** in increments of 25	Enter the minimum interval between the receiving switch's seizure acknowledgment and the outpulsing of digits by this switch.

Incoming Dial Guard (msec)

Valid entries	Usage
10 to **2550** in increments of 10	Enter the minimum acceptable interval between the detection of an incoming seizure and the acceptance of the first digit. The switch will not accept digits before this timer expires.

NOTE:
This timer is never sent to TN429 ports.

Outgoing Dial Guard (msec)

Valid entries	Usage
100 to **25500** in increments of 100	Enter the minimum interval between seizure acknowledgment of a trunk and the outpulsing of digits. This field cannot be blank. For trunks that do not provide seizure acknowledgment, the timer specifies the minimum time between seizure and the outpulsing of digits. Any digit the caller dials after they lift the receiver, but before the timer expires, is not outpulsed until the timer expires.

Incoming Glare Guard (msec)

This field only appears when the trunk group Direction field is **two-way**.

Valid entries	Usage
100 to **25500** in increments of 100	Enter the minimum interval that must elapse between a trunk's release from an incoming call and its seizure for an outgoing call. This field cannot be blank. This delay gives the far-end time to release all equipment after the trunk is released.

Outgoing Glare Guard (msec)

This field only appears for **outgoing** and **two-way** trunk groups.

Valid entries	Usage
100 to **25500** in increments of 100	Enter the minimum interval that must elapse between a trunk's release from an outgoing call and its seizure for another outgoing call. This field cannot be blank. This delay gives the far-end time to release all equipment after the outgoing trunk is released.

Outgoing Rotary Dial Interdigit (msec)

This field only appears when: (1) the trunk group Group Type field is **access**, **aplt**, **co**, **diod**, **dmi-bos**, **fx**, **rlt**, **tandem**, or **wats** and the Outgoing Dial Type field is **rotary**, (2) the Group Type field is **tie**, the Trunk Signaling Type field is blank, **cont**, or **dis**, and the Outgoing Dial Type field is **rotary**, or (3) the Group Type field is **tie**, and the Trunk Signaling Type field is **tge**, **tgi**, or **tru** (the Outgoing Dial Type field does not appear but is implied to be **rotary**).

Valid entries	Usage
150 to **2550** in increments of 10	Enter the minimum time between outpulsed digits on outgoing rotary trunks.

Ringing Monitor (msec)

This timer is sent to TN464C (or later), TN767, TN438 (all), TN447, TN465 (all), TN2138, TN2147, TN2184, and TN2199 CO circuit packs.

Valid entries	Usage
200 to **51000** in increments of 200	Enter the minimum time the switch requires to determine if a trunk disconnects. The field cannot be blank. If the ringing signal disappears for a duration longer than the time specified in this field, the switch assumes the call has been disconnected.

Cama Wink Start Time (msec)

Valid entries	Usage
20 to **5100** in increments of 20	Specifies the duration (the wait-for-wink-to-end time) for a wink-start CAMA trunk. The wink must begin before the Outgoing Seizure Response timer expires.

Incoming Partial Dial (sec)

This timer appears only if the Incoming Dial Type field is **rotary**.

Valid entries	Usage
5 to **255** in increments of 1	Enter the maximum time allowed between incoming rotary digits.

NOTE:
This timer is never sent to TN429 ports.

Incoming Seizure (msec)

This field appears when the Direction field is **incoming** or **two-way,** and, when applicable, the Trunk Signaling Type field is **cont**. Only TN429, TN438 (any release), TN 447, TN464C (or later), TN465 (any release), TN767, TN2138, TN2140, TN2147, TN2184, and TN2199 ports receive this timer. For DID trunks, only TN2199 and TN429D (or later) receive this timer.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of the shortest incoming seizure signal your switch can recognize. For ICLID, set this field to 120. The field cannot be blank.

Outgoing Rotary Dial Interdigit (msec)

This field only appears when the Outgoing Dial Type field is **rotary**.

Valid entries	Usage
150 to **2550** in increments of 10	Enter the minimum acceptable interval between outgoing rotary digits. The field cannot be blank.

Outgoing End of Dial (sec)

This field controls firmware answer supervision timers on circuit packs that have them. It appears when the Direction field is **outgoing** or **two-way** and the Receive Answer Supervision field is **n**. If the Receive Answer Supervision field is **y**, this field does not appear and the firmware timer on the appropriate circuit pack is automatically disabled.

NOTE:
The Answer Supervision Timeout field on the Trunk Group screen provides timed answer supervision for circuit packs without administrable timers. Since trunk groups may contain ports on more than one circuit pack, it's possible you may need to use both timers with the same trunk group. If so, set the Outgoing End of Dial field and the Answer Supervision Timeout field to the same value.

Valid entries	Usage
1 to **254** in increments of 1	Enter the maximum time, in seconds, that the switch will wait to receive answer supervision for outgoing calls on the ports controlled by firmware timers. For Brazil pulsed E&M signaling, use **40**.

During a cut-through operation, timing begins after the switch sends each outgoing digit and ceases when answer supervision is received. If the timer expires, the switch acts as if it has received answer supervision. On senderized operation, the timer begins after the switch sends the last digit collected. The timer ceases when answer supervision is received. If the timer expires, the switch acts as if it has received answer supervision.

Outgoing Seizure Response (sec)

This timer is sent to the TN438B, TN439, TN447, TN458, TN464B (or later), TN465B (or later), TN767, TN2140, TN2147, TN2184, TN2199, and TN2242 circuit packs.

Valid entries	Usage
1 to **255** in increments of 1	Enter the maximum interval that the switch should wait after sending a seizure signal to receive seizure acknowledgment from the far-end. If the acknowledgment is not received in this time, a seizure failure response is uplinked. For Brazil pulsed E&M signaling, use **255**.

Programmed Dial Pause (msec)

This timer is administrable for all outgoing and two-way trunk groups. This timer works with the TN464B (or later), TN767, TN458, TN2140, and TN2242 tie circuit packs. All CO circuit packs that accept administrable timers accept this timer.

Valid entries	Usage
100 to **25500** in increments of 100	Set the exact duration of pauses used during abbreviated dialing, ARS outpulsing, and terminal dialing operations.

Disconnect Signal Error (sec)

This field appears for ground-start trunk groups.

Valid entries	Usage
1 to **255** in increments of 1	Enter the maximum interval that the switch will wait to receive a disconnect signal from the far-end switch after the local party (a phone or tie trunk) goes on-hook. If the timer expires, the system assumes a disconnect failure and take appropriate action such as creating an error message.

Flash Length (msec)

This timer is sent to TN436B, TN459B, TN464C (or later), TN465B (or later), (TN753 if Country is **23**), TN2146, TN2147, TN2184, and TN2199 circuit boards.

Valid entries	Usage
10 to **2550** in increments of 10	Enter the duration of a flash signal generated toward the central office.

Busy Tone Disconnect

The field appears when Enable Busy Tone Disconnect for Analog loop-start Trunks is **y** on the System Parameters Country-Options screen.

Valid entries	Usage
y/n	Enter **y** to allow the switch to recognize a busy tone signal as a disconnect on this trunk group.

Outgoing Seizure (msec)

Appears when the Country field is **15**, the Direction field is **outgoing** or **two-way**, and the Trunk Type field is **2-wire-ac**, **2-wire-dc**, or **3-wire**. This timer is sent only to the TN2199 circuit pack.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of the outgoing seizure signal.

Incoming Incomplete Dial Alarm (sec)

Only the TN436 (all), TN459 (all), TN464C (or later), TN767, TN2140, TN2146, TN2184, TN2199, and TN2242 circuit packs use this timer.

Valid entries	Usage
1 to **255** in increments of 1	Enter the maximum acceptable interval between an incoming seizure and receipt of all digits. Intervals greater than this limit generate an inline error.

Outgoing Last Digit (sec)

This field is only administrable if the Trunk Signaling Type field is **dis** or **cont** and the trunk group Direction field is **two-way** or **outgoing**. Only TN497 and TN2140 ports receive this timer.

Valid entries	Usage
1 to **40**	Enter the maximum time that the switch will wait for the next digit dialed. After the timer expires, no more digits are accepted by the circuit pack.

Glare

This field is only administrable if the Trunk Signaling Type field is **cont** and the trunk group Direction field is **two-way** or **outgoing**. Only TN2140 ports receive this timer.

Valid entries	Usage
40 to **100** in increments of 10	Enter the minimum acceptable interval (in msec) between the moment your switch sends an outgoing seizure and the moment it receives a seizure acknowledgment. If acknowledgment is received before the timer expires, glare is assumed.

Normal Outgoing Seize Send (msec)

This field appears only if the Trunk Signaling Type field is **dis** and the trunk group Direction field is **two-way** or **outgoing**. Only TN2140 ports receive this timer.

Valid entries	Usage
10 to **990** in increments of 10	Enter the duration of the signal your switch sends for an outgoing seizure.

Release Ack Send (msec)

After your switch receives a forward release signal, it must send a forward release acknowledgment signal. This field appears only if the Trunk Signaling Type field is **dis** and the trunk group Direction field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
500 to **1200** in increments of 100	Enter the duration of the signal your switch sends for a forward release acknowledgment.

Seize Ack Delay (msec)

This field appears only if the Trunk Signaling Type field is **dis** and the trunk group Direction field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
40 to **120** in increments of 10	Enter the maximum interval your switch will wait after receipt of an incoming seizure to send seizure acknowledgment.

Answer Send (msec)

This field appears only if the Trunk Signaling Type field is **dis** and the trunk group Direction field is **incoming** or **two-way**. Only TN2140 and TN2199 ports receive this timer.

Valid entries	Usage
10 to **2550** in increments of 10	Enter the duration of the answer signal pulse.

Outgoing Disconnect Send (msec)

This field is administrable only if the Trunk Signaling Type field is **dis** and the trunk group Direction field is **two-way** or **outgoing**. Only TN2140 ports receive this timer.

Valid entries	Usage
100 to **9900** in increments of 100	Enter the duration of the forward release signal your switch sends at the end of outgoing calls.

Seize Ack Send (msec)

This field appears only if the Trunk Signaling Type field is **dis** and the trunk group Direction field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
10 to **990** in increments of 10	Enter the duration of the seizure acknowledgment signal your switch sends in response to an incoming seizure.

Incoming Disconnect Send (msec)

This field is only administrable if the Trunk Signaling Type field is **dis** and the trunk group Direction field is **incoming** or **two-way**. Only TN2140 ports receive this timer.

Valid entries	Usage
500 to **1200** in increments of 100	Enter the duration of the backward release signal your switch sends at the end of an incoming call.

Tone (msec)

This field appears only if the Trunk Signaling Type field is blank. All CO, DIOD, and Tie circuit packs that accept administrable timers accept this timer. This timer is also sent to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, TN429, TN2184 ports in a DID trunk group.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the duration of a DTMF tone sent when a button on a hybrid phone is pressed.

Pause (msec)

THis field is administrable only if the Trunk Signaling Type field is blank. All CO, DIOD, and tie circuit packs that accept administrable timers accept this timer. However, this timer is sent only to the following circuit packs: TN464B (or later), TN767, TN436B, TN459B, TN2146, TN2199, and TN2242, and TN429 and TN2184 ports in a DID trunk group.

Valid entries	Usage
20 to **2550** in increments of 10	Enter the minimum acceptable interval (pause) between DTMF tones sent from a hybrid phone.

PPS

Valid entries	Usage
10 **20**	Enter the rate (pulses per second) at which outgoing rotary pulses are sent over this trunk group.

NOTE:
The TN439, TN458, TN497, TN747Bv12 (or later), and TN767 circuit packs only send rotary pulses at 10 pps.

Make (msec)

Valid entries	Usage
	Enter the duration of the make interval (the pause between pulses) while the system is outpulsing digits using dial pulse signaling. The field cannot be blank.
20 to **80** in increments of 5	If the PPS field is **10**, the sum of the Make (msec) and Break (msec) fields must equal 100.
10 to **40** in increments of 5	If the PPS field is **20**, the sum of the Make (msec) and Break (msec) fields must equal 50.

Break (msec)

Valid entries	Usage
	Enter the duration of the break interval (the pulse duration) while the system is outpulsing digits using dial pulse signaling. The field cannot be blank.
20 to **80** in increments of 5	If PPS field is **10**, the sum of the Make (msec) and Break (msec) fields must equal 100.
10 to **40** in increments of 5.	If the PPS field is **20**, the sum of the Make (msec) and Break (msec) fields must equal 50.

PPM

For CO, DIOD, FX, PCOL, and WATS trunks. This field appears when the Direction field is **outgoing** or **two-way**.

Valid entries	Usage
y/n	Enter **y** if Periodical Pulse Metering (PPM) pulses should be collected from the public network to determine call cost. If this field is **y**, the Frequency field appears.

Frequency

This field identifies the PPM pulse frequency(ies) sent by the public network. It appears if the Direction field is **outgoing** or **two-way** and PPM is **y**. Circuit packs can detect up to three different frequencies, (12kHz, 16kHz, and 50Hz), plus two frequency combinations, (50Hz/12kHz and 50Hz/16kHz). This field controls TN465B, TN2138, and TN2184 circuit packs.

Valid entries	Usage
12k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **12k** is administered, the circuit pack will be set to detect 12kHz.
16k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **16k** is administered, the circuit pack will be set to detect 16kHz.
50	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **50** is administered, the circuit pack will be set to detect 16kHz.
50/12k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **50/12k** is administered, the circuit pack will be set to detect 12kHz.
50/16k	The TN465B (or later) and TN2184 can only detect 12k and 16kHz PPM. Therefore, if **50/16k** is administered, the circuit pack will be set to detect 16kHz.

Field descriptions for page 4

This screen appears when the Direction field on Page 1 is **outgoing** or **two-way** and the ATMS field is **y** on the Feature-Related System Parameters screen.

The figure below shows a common configuration for page 4 of the Trunk Group screen when the Group Type field is **co**. This screen is only an example, and the fields shown below may change or disappear according to specific field settings.

```
add trunk-group next                                                   Page 4 of x
                          ATMS THRESHOLDS
TTL Type: _________                  Far End Test No: __________________
  TTL Vendor: ______________             TTL Contact: ______________________
Trunk Vendor: ____________             Trunk Contact: ___________________
Trunk Length: _____
                         MARGINAL         UNACCEPTABLE
                         Min   Max           Min   Max
           1004 Hz Loss: __     __           __    __

                         -Dev  +Dev          -Dev   +Dev
            404 Hz Loss:   _     _             _      _
           2804 Hz Loss:   _     _             _      _

Maximum C Message Noise:     __                  __
Maximum C Notched Noise:     __                  __
         Minimum SRL-HI:     __                  __
         Minimum SRL-LO:     __                  __
            Minimum ERL:     __                  __

      Allow ATMS Busyout, Error Logging and Alarming? _
  Maximum Percentage of Trunks Which Can Be Removed from Service by ATMS: __
```

Screen 268. CO Trunk Group ATMS Thresholds

CAUTION:
Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

TTL Type

Specifies the type of terminating test line (TTL) selected for testing trunks. The TTL type determines what ATMS tests can be completed and thus which threshold values need to be administered.

Valid entries	Usage
105-w-rl	105 with return loss
105-wo-rl	105 without return loss
high-lts	high-level tone source
low-lts	low-level tone source
100	100 type
102	102 type

The following table explains the differences between types of terminating test lines:

Type TTL	Description	Example
105-w-rl	Full range of 18 measurements or some defaults for return loss used (56A)	TN771B, ZLC12 and SN261B circuit packs and new 56A mini-responder
105-wo-rl	Cannot return default values for far-end return loss	Older 56A mini-responder
high-level-tone	Sends a fixed sequence of tones at 0 dBm	ZLC12 and SN261B circuit packs
low-level-tone	Sends a fixed sequence of tones at -16dBm	SN261B circuit pack
100	Up to 5 measurements that sends a 1004 Hz tone then a quiet termination	
102	One measurement that sends a 1004 Hz tone	

The far-end switch containing the TTL may be any of the following:

- System 85 R2 switch, equipped with the Maintenance/Test Board (TN771B)
- System 75 R1V2 and beyond, all of which contain the circuitry required to perform the TTL function
- System 85 R2 switch, equipped with the Analog/Digital Facility Test Circuit (ADFTC, SN261)
- DIMENSION FP8, equipped with the Analog Facility Test Circuit (AFTC, ZLC-12)
- Central Office switches, equipped with various TTL equipment that provide 100, 102, or 105 test line capabilities (56A)

Other vendors’ switches *may* be supported if compatible test lines are provided by these switches.

Four different versions of the ATMS Threshold Administration page can appear depending upon the measurements allowed by the TTL type selected. The four possibilities are:

1. 105-w-rl and 105-wo-rl — All thresholds appear.
2. high-lts and low-lts — All thresholds (except maximum C-notched noise) appear.
3. 100 — All thresholds (except maximum c-notched noise, 404Hz loss, and 2804 Hz loss) appear.
4. 102 — Only 1004 Hz loss threshold appears.

Far-End Test No.

Valid entries	Usage
1 to 16 digits	Enter the access number dialed to reach the terminating test line (TTL).

TTL Vendor

Valid entries	Usage
0 to 22 alphanumeric characters	Enter the name of the vendor supplying the terminating test line (TTL).

TTL Contact

Valid entries	Usage
0 to 25 alphanumeric characters	Enter the name and/or telephone number of someone from the TTL vendor who can be contacted in the event of problems with the terminating test line.

Trunk Vendor

Valid entries	Usage
0 to 22 alphanumeric characters	Enter the name of the vendor providing service over this trunk group (the company to notify in the event of problems with the trunks in this trunk group).

Trunk Contact

Valid entries	Usage
0 to 25 alphanumeric characters	Enter the name and/or telephone number of someone from the trunk vendor who can be contacted in the event of problems with the trunks.

Trunk Length

This field is not required. Since noise on a trunk increases with the length of the trunk, however, this information may be useful,

Valid entries	Usage
Use this field to record the length of the trunk group in kilometers or miles.	
0–4 digits followed by **k**	Shows the length in kilometers.
0–4 digits followed **m**	Shows the length in miles.

Marginal Threshold - Min -1004 Hz Loss

Valid entries	Usage
-2 to **21**	Enter the minimum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as out of tolerance. A larger dB value is more restrictive.

Marginal Threshold - Max - 1004 Hz Loss

Valid entries	Usage
0 to **21**	Enter the maximum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as out of tolerance. A smaller dB value is more restrictive.

Unacceptable Threshold - Min - 1004 Hz Loss

Valid entries	Usage
-2 to **21**	Enter the minimum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as unacceptable. A larger dB value is more restrictive.

Unacceptable Threshold - Max - 1004 Hz Loss

Valid entries	Usage
0 to **21**.	Enter the maximum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as unacceptable. A smaller dB value is more restrictive.

Marginal Threshold - -Dev - 404 Hz Loss

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 404 Hz from the 1004 Hz test tone noise level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - +Dev - 404 Hz Loss

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Unacceptable Threshold - -Dev - 404 Hz

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - +Dev - 404 Hz

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Marginal Threshold - -Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - +Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Unacceptable Threshold - -Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum negative deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - +Dev - 2804 Hz

Valid entries	Usage
0 to **9**	Enter the maximum positive deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Marginal Threshold - Maximum C Message Noise

Valid entries	Usage
15 to **55**	Enter the maximum C-message noise phone as measured within the voice band frequency range (500 to 2500 Hz) allowed before reporting a trunk as out of tolerance. Smaller values are more restrictive.

Unacceptable Threshold - Maximum C Message Noise

Valid entries	Usage
15 to **55**	Enter the maximum C-message noise interference in dBmC above reference noise terminating on a phone as measured within the voice band frequency range (500 to 2500 Hz) allowed before reporting a trunk as unacceptable. Smaller values are more restrictive.

Marginal Threshold - Maximum C Notched Noise

Valid entries	Usage
34 to **74**	Enter the maximum C-notched signal dependent noise interference in dBmC allowed before reporting a trunk as out of tolerance. Smaller values are more restrictive.

Unacceptable Threshold - Maximum C Notched Noise

Valid entries	Usage
34 to **74**	Enter the maximum C-notched signal dependent noise interference in dBmC allowed before reporting a trunk as unacceptable. Smaller values are more restrictive.

Marginal Threshold - Minimum SRL-HI

Valid entries	Usage
0 to **40**	Enter the minimum high-frequency signaling return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Unacceptable Threshold - Minimum SRL-HI

Valid entries	Usage
0 to **40**	Enter the minimum high-frequency signaling return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Marginal Threshold - Minimum SRL-LO

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency signaling return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Unacceptable Threshold - Minimum SRL-LO

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency signaling return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Marginal Threshold - Minimum ERL

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency echo return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Unacceptable Threshold - Minimum ERL

Valid entries	Usage
0 to **40**	Enter the minimum low-frequency echo return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Allow ATMS Busyout, Error Logging and Alarming

Valid entries	Usage
y/n	Enter **y** to allow ATMS error logging and alarming (subject to filtering depending on the service organization used to deal with alarms).

Maximum Percentage of Trunks Which Can Be Removed From Service by ATMS

Valid entries	Usage
0 to **100**	Enter the highest percentage of trunks from the trunk group that can be removed from service at one time because of unacceptable transmission measurement results.

Field descriptions for page 5

The total number of pages, and the first page of Group Member Assignments, vary depending on whether the Administrable Timers and ATMS Thresholds pages display.

```
add trunk-group next
                                 TRUNK GROUP
                                        Administered Members(min/max): xxx/yyy
                                           Total Administered Members: xxx
GROUP MEMBER ASSIGNMENTS
       Port     Code  Sfx  Name         Night          Mode     Type   Ans Delay
    1: _______             ___________  _____          ________ _______ ____
    2: _______             ___________  _____          ________ _______ ____
    3: _______             ___________  _____          ________ _______ ____
    4: _______             ___________  _____          ________ _______ ____
    5: _______             ___________  _____          ________ _______ ____
    6: _______             ___________  _____          ________ _______ ____
    7: _______             ___________  _____          ________ _______ ____
    8: _______             ___________  _____          ________ _______ ____
    9: _______             ___________  _____          ________ _______ ____
   10: _______             ___________  _____          ________ _______ ____
   11: _______             ___________  _____          ________ _______ ____
   12: _______             ___________  _____          ________ _______ ____
   13: _______             ___________  _____          ________ _______ ____
   14: _______             ___________  _____          ________ _______ ____
   15: _______             ___________  _____          ________ _______ ____
```

Screen 269. Group Member Assignments

Administered Members (min/max)

This display-only field shows the minimum and maximum member numbers that have been administered for this trunk group.

Total Administered Members

This display-only field shows the total number of members administered in the trunk group.

Port

If this trunk is registered as an endpoint in an IP application, this field will display T00000.

Valid entries	Usage
a valid port ID on a trunk circuit pack	Enter the port number of each member. The member number of the trunk is the number displayed to the left of the Port field.
IP	For trunk members of trunk groups with the Carrier Medium field set to **IP**.

NOTE:
In DCS networks, trunks must be assigned the same member number at both nodes.

NOTE:
Members assigned to IP trunk groups will display a value of “T00001.”

Code

This display-only field shows the type of circuit pack physically installed or logically administered at the location to which this member is assigned. If no circuit pack is installed or administered at the port address you enter, the field is blank.

Sfx

This display-only field shows the model suffix for the type of circuit pack physically installed at the location to which this member is assigned. If no circuit pack is installed at the port address you enter, the field is blank.

Name

Your vendor, as well as Avaya technical staff, sometimes need to identify specific trunks to work with your system. Therefore, the name you give to a trunk should identify the trunk unambiguously.

Valid entries	Usage
Up to 10 characters	Examples of good names: ■ The phone number assigned to incoming trunks ■ The Trunk Circuit Identification number assigned by your service provider

Night

Use this field only if you want to assign a night service destination to individual trunks that is different from the group destination entered in the Night Service field on page 1. Incoming calls are routed to this destination when the system is placed in night service mode.

Valid entries	Usage
a valid extension	Enter the extension of the night destination for the trunk.
attd	Enter **attd** if you want calls to go to the attendant when night service is active.

Mode

This field specifies the signaling mode used on tie trunks with TN722A or later, TN760B or later, TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs. This entry must correspond to associated dip switch settings on the circuit pack.

CAUTION:
Customers should not attempt to administer this field. Please contact your Avaya representative for assistance.

Valid entries	Usage
e&m	Enter **e&m** for 6-wire connections that pair 2 signaling wires with 4 voice wires. You'll use **e&m** in the vast majority of systems in the U.S.
simplex	Enter **simplex** for 4-wire connections that do not use an additional signaling pair. This configuration is very rare in the U.S.
protected	

Type

This field specifies the signaling type to be used with TN760B (or later release), TN722 (with any suffix), TN767, TN2140 (when the Trunk Signaling Type field is **cont**), TN437, TN439, TN464 with any suffix, or TN458 circuit packs.

The Type column appears when the Trunk Signaling Type field is blank or **cont**. The Type column does not display if the Trunk Signaling Type field is **dis**

CAUTION:
Customers should not attempt to administer this field. Please contact your Avaya representative for assistance.

Valid entries	Usage
t1 stan	
t1 comp	
t5rev	The value of **t5 rev** is allowed only for the TN760D vintage 10 or later. When Type is **t5 rev**, Mode must be **e&m**.
type 5	

Ans Delay

CAUTION:
Customers should not attempt to administer this field. Please contact your Avaya representative for assistance

Valid entries	Usage
20 to **5100** in increments of 20	Specifies the length of time (in ms) your switch will wait before it sends answer supervision for incoming calls on tie trunks using the TN722A or later, TN760 (B, C, or D), TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs.
blank	Same as setting the field to zero.

This delay serves two purposes:

- It ensures that the answer supervision signal is valid and not a secondary delay-dial or wink-start signal.
- It ignores momentary off-hook signals resulting from connections made off-network through certain No. 5 Crossbar CCSA switches as the connection is being established. Therefore, calls aren't dropped inappropriately.

Related topics

Refer to ''Managing trunks'' on page 403 for instructions on adding and managing trunk groups.

Refer to ''Trunks and Trunk Groups'' on page 1782 for detailed information about all types of trunk groups except ISDN.

Refer to ''ISDN service'' on page 1610 for detailed information on Integrated Services Digital Network.

Uniform Dial Plan Table

The Uniform Dialing Plan field must be y on the System-Parameters Customer-Options screen before you can administer this table.

The UDP provides a common 3- to 7-digit dial plan length — or a combination of extension lengths — that can be shared among a group of switches. Additionally, UDP can be used alone to provide uniform dialing between two or more private switching systems without ETN, DCS, or Main/Satellite/Tributary configurations.

See “Uniform Dial Plan” on page 1792 for more information on Uniform Dial Plans.

```
change uniform-dialplan 0                                           Page 1 of 2
                                UNIFORM DIAL PLAN TABLE
                                                         Percent Full:   2

Matching             Insert          Node      Matching            Insert          Node
Pattern   Len Del    Digits Net Conv Num       Pattern   Len Del   Digits Net Conv Num
2_____     4   0     817_   aar  n   __        _____     __  __    ___    __  __   __
4_____     5   1     334_   aar  n   __        _____     __  __    ___    __  __   __
43659      5   1     928_   aar  y   __        _____     __  __    ___    __  __   __
623___     3   3     5380   aar  n   __        _____     __  __    ___    __  __   __
73012      5   1     ____   enp  n   31        _____     __  __    ___    __  __   __
74100      5   0     81__   ars  y   __        _____     __  __    ___    __  __   __
8_____     5   0     ____   ext  n   __        _____     __  __    ___    __  __   __
911___     3   0     ____   ars  n   __        _____     __  __    ___    __  __   __
_____      _   _     ____   ___  _   __        _____     __  __    ___    __  __   __
_____      _   _     ____   ___  _   __        _____     __  __    ___    __  __   __
_____      _   _     ____   ___  _   __        _____     __  __    ___    __  __   __
_____      _   _     ____   ___  _   __        _____     __  __    ___    __  __   __
_____      _   _     ____   ___  _   __        _____     __  __    ___    __  __   __
_____      _   _     ____   ___  _   __        _____     __  __    ___    __  __   __
```

Screen 270. Uniform Dial Plan Table

Percent Full

Displays the percentage (0 to 100) of the memory resources allocated for the uniform dial plan data that are currently being used.

Matching Pattern

Valid entries	Usage
0–9 (1 to 7 digits)	Enter the number you want the switch to match to dialed numbers.

Len

Valid entries	Usage
3–7	Enter the minimum number of user-dialed digits the system collects to match to this Matching Pattern. This number must be greater than or equal to the number entered in the Matching Pattern field.

Del

Valid entries	Usage
0–3	Enter the number of digits to delete before routing the call. This number must be less than or equal to the number entered in the Len field.

Insert Digits

Valid entries	Usage
0 to **9** (1 to 4 digits)	Enter the digits that replace the deleted portion of the dialed number. Leave this field blank to simply delete the digits.

Net

Enter the switch network used to analyze the converted number.

Valid entries	Usage
aar, ars, enp, ext	The converted digit-string will be routed either as an extension number or via its converted AAR address, its converted ARS address, or its ENP node number. If you enter **enp**, you must enter the ENP node number in the Node Num field. The Insert Digits field must be blank, and Conv must be **n**.

Conv

Valid entries	Usage
y/n	Enter y to allow additional digit conversion

Node Num

This is the ENP (Extension Number Portability) Node Number.

Valid entries	Usage
1–999	Enter the ENP node number.

Vector Directory Number

This screen defines vector directory numbers (VDN) for the Call Vectoring feature. A VDN is an extension number used to access a call vector. Each VDN is mapped to one call vector.

VDNs are software extension numbers (that is, not assigned to physical equipment). A VDN is accessed via direct dial CO trunks mapped to the VDN (incoming destination or night service extension), DID trunks, and LDN calls. The VDN may be Night Destination for LDN.

Refer to the *Avaya MultiVantage Call Vectoring/Expert Agent Selection (EAS) Guide* for more information.

Field descriptions for page 1

```
change vdn 5000                                                    Page  1 of   1
                       VECTOR DIRECTORY NUMBER

                             Extension: 5000
                                  Name:
                         Vector Number: 234
                   Attendant Vectoring: n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 59
                                    TN: 1
                              Measured: none
        Acceptable Service Level (sec):
         VDN of Origin Annc. Extension: 301
                             1st Skill:
                             2nd Skill:
                             3rd Skill:
```

Screen 271. Vector Directory Number

Extension

This is a display-only field when using an administration command such as **add** or **change** to access the screen. The extension is a 1- to 7-digit number that starts with a valid first digit and length as defined by the System's dial plan.

Name

This is an optional field that need not contain any data. It is the name associated with the VDN.

Valid entries	Usage
Enter up to a 27-character alphanumeric name that identifies the VDN.	The name may be truncated on agents' displays depending on the application. When information is forwarded with an interflowed call, only the first 15 characters are sent

Vector Number

Identification number that specifies a particular call vector that is accessed through the VDN.

Valid entries	Usage
1 to **256**	Enter a 1- to 3-digit vector number. This field cannot be blank.

Attendant Vectoring

This field appears only if, on the System-Parameters Customer-Options screen, the Attendant Vectoring is **y**. This field identifies if the vector you are defining is an attendant vectoring VDN.

Valid entries	Usage
y	Enter **y** so the vector is an attendant vector. This entry will dynamically change the rest of the screen to eliminate field options available with other types of vectors. Default when, on the System-Parameters Customer-Options screen, both the Vectoring (Basic) and Vectoring (Prompting) fields are **n**. No changes allowed to this field.
n	Default when, on the System-Parameters Customer-Options screen, the Vectoring (Basic) and/or Vectoring (Prompting) fields are **y**.

Meet-me Conference

This field appears only if, on the System-Parameters Customer-Options screen, the Enhanced Conferencing field is **y**. This field determines if the VDN is a Meet-me Conference VDN.

NOTE:
If the VDN extension is part of your DID block, external users will be able to access the conference VDN. If the VDN extension is not part of your DID block, only internal callers on the your network (including DCS or QSIG) or remote access callers can access the conference VDN.

Valid entries	Usage
y/n	Enter **y** to enable Meet-me Conference for this VDN. If Meet-me Conference is **y**, only Extension, Name, Vector Number, Meet-me Conference, COR, and TN fields display and the fields for page 2 change. Both Attendant Vectoring and Meet-me Conference cannot be enabled at the same time. If Enhanced Conferencing is **y**, but no other vectoring options are enabled, only Meet-me Conference vectors can be assigned.

Allow VDN Override

This entry affects the operation of an agent’s display and certain options/data assigned to the VDN when a call is routed through several VDNs.

For Expert Agent Selection (EAS), if this field is **y** on the original VDN, the Skills of the new VDN will be used. If this field is **n** on the original VDN, the Skills of the original VDN will be used.

For Best Service Routing (BSR), if this field is **y** on the original VDN, the BSR Application and Available Agent Strategy of the new VDN will be used. If this field is **n** on the original VDN, the BSR Application and Available Agent Strategy of the original VDN will be used.

Valid entries	Usage
y	The name of the VDN appearing on the terminating display depends on the administration and chaining of the subsequent VDNs and the AUDIX mail for the last VDN is accessed.
n	The name of this VDN appears on the agent’s display and the VDN’s AUDIX mail is accessed. If any subsequent VDNs are used to process this call, their names will not appear on the terminating display and the AUDIX mail for the original VDN is accessed.

COR

Specifies the class of restriction (COR) of the VDN.

Valid entries	Usage
0 to **95**	Enter a 1- or 2-digit number. This field cannot be blank.

TN

Specifies the Tenant Partition number for this VDN.

Valid entries	Usage
1 to **20**	

Measured

Used to collect measurement data for this VDN. Data may be collected for reporting by BCMS or CMS.

NOTE:
On the System-Parameters Customer-Options screen, the BCMS field must be **y** for the Measured field to be set to **internal** or **both**. In addition, the appropriate CMS release must be administered on the Feature-Related System Parameters screen if this field is being changed to **external** or **both**.

Valid entries	Usage
internal	Data will be measured internally by BCMS.
external	Data will be measured internally by CMS.
both	Data will be measured internally by both BCMS and CMS.
none	Data will not be measured.

Acceptable Service Level (sec)

Only appears when, on the System-Parameters Customer-Options screen, the BCMS/VuStats Service Level field is **y** and the Measured field is **internal** or **both**.

Valid entries	Usage
0 to **9999** seconds	Enter the number of seconds within which calls to this VDN should be answered. This will allow BCMS to print out a percentage of calls that were answered within the specified time.

Service Objective

Appears when Skill and Centre Vu Advocate are **y** on the Feature Related System Parameters screen.

Valid entries	Usage
1-9999	Enter the service objective.

VDN of Origin Annc. Extension

Data for this field appears only when, on the System-Parameters Customer-Options screen, the VDN of Origin Announcement field is **y**.

Valid entries	Usage
VDN extension	Enter the extension number of the VDN of Origin announcement.

1st/2nd/3rd Skill

Only appears when, on the System-Parameters Customer-Options screen, the Expert Agent Selection (EAS) field is **y**.

Valid entries	Usage
1 to **99**	Enter the desired Skill numbers in each field.

Field descriptions for page 2

```
change vdn 5000                                                   Page  2 of   2
                       VECTOR DIRECTORY NUMBER

                            AUDIX Name:
                 Messaging Server Name:
                    Return Destination:
                VDN Timed ACW Interval:
                       BSR Application:
          BSR Available Agent Strategy: 1st-found
               Observe an Agent Answer?
                Conference Access Code:
                 Conference Controller:
          Display VDN for Route-To DAC?
```

Screen 272. Vector Directory Number

AUDIX Name

Only appears for DEFINITY Server R configurations. If this VDN is associated with the AUDIX vector, enter the name of the AUDIX machine as it appears in the Node Names screen.

Messaging Server Name

Only appears for R5r configurations.

Return Destination

Valid entries	Usage
VDN extension	Enter the VDN extension number to which an incoming trunk call will be routed if it returns to vector processing after the agent drops the call.

VDN Timed ACW Interval

When a value is entered in this field, an agent in auto-in work mode who receives a call from this VDN is automatically placed into After Call Work (ACW) when the call drops. When the administered time is over, the agent automatically becomes available. This field has priority over the Timed ACW Interval field on the Hunt Group screen.

Valid entries	Usage
1-9999 or blank	Enter the number of seconds the agent should remain in ACW following the call.

BSR Application

To use multi-site Best Service Routing with this VDN, specify an application plan for the VDN. This field only appears if, on the System Parameters Customer-Options screen, the Lookahead Interflow (LAI) and Vectoring (Best Service Routing) fields are **y**.

Valid entries	Usage
1-255 or blank	Enter a 1- to 3-digit number.

BSR Available Agent Strategy

The available agent strategy determines how Best Service Routing identifies the "best" split or skill to service a call in an agent surplus situation. To use Best Service Routing with this VDN, enter an agent selection strategy in this field.

This field only appears if, on the System Parameters Customer-Options screen, the Vectoring (Best Service Routing) field is **y**.

Valid entries	Usage
1st-found	
UCD-LOA	
UCD-MIA	
EAD-LOA	
EAD-MIA	

Observe an Agent Answer

This field allows for a service observer to start observing a call to the VDN when the call is delivered to the agent/station.

Valid entries	Usage
y/n	

Conference Access Code

This field appears only if Meet-me Conference is **y**. This field allows you to assign an access code to the Meet-me Conference VDN.

SECURITY ALERT:

You should always assign an access code to a Meet-me Conference VDN.

Valid entries	Usage
blank or 6-digit number	Enter a 6-digit access code for the Meet-me Conference VDN. If you do not want an access code, leave the field blank. Once an access code is assigned, an asterisk displays in this field for subsequent change, display, or remove operations by all users except the "init" superuser login.

Conference Controller

This field appears only if Meet-me Conference is **y**. This field controls which user is allowed to change the access code for a Meet-me Conference VDN using a feature access code. This can be a local user or someone dialing in via remote access trunks.

Valid entries	Usage
extension number or blank	If an extension number is entered, only a user at that extension can change the access code for that VDN using a feature access code. If this field is blank, any station user that is assigned with console permissions can change the access code for that VDN using a feature access code.

Display VDN for Route-To DAC

This field addresses situations where one of the following conditions is in effect:

- A **route-to number** or **route-to digits** vector command routes an EAS direct agent call, with coverage option set to **y**
- An adjunct routing step routes a direct agent call, with the coverage option set to **y**

For more information, see *Avaya MultiVantage™ Call Center Software Call Vectoring and Expert Agent Selection (EAS) Guide*, 585-230-714.

Valid entries	Usage
y/n	

VDN Override for ISDN Trunk ASAI Messages

This field appears only if, on the System-Parameters Customer-Options screen, the ASAI Link Core Capabilities field is **y** and if the Allow VDN Override field on this screen is **y**. This field allows those whose CTI applications require the Called Number ASAI event information to correspond to the active VDN for an incoming ACD call rather than the Called Number information in the incoming ISDN call's SETUP message.

Field descriptions for page 2

```
change vdn 5000                                              Page  2 of 2    SPE A
                         VECTOR DIRECTORY NUMBER

                      MEET-ME CONFERENCE PARAMETERS
    Conference Access Code:
     Conference Controller:
```

Screen 273. Vector Directory Number (if the Meet-me Conference field is y)

Conference Access Code

This field appears only if the Meet-me Conference field is **y**.

SECURITY ALERT:

You should always assign an access code to a Meet-me Conference VDN.

Valid entries	Usage
6-digit number or blank	Enter a 6-digit access code for the Meet-me Conference VDN. If you do not want an access code, leave blank. Once an access code is assigned, an asterisk displays in this field for subsequent change, display, or remove operations by all users except the “init” superuser login.

Conference Controller

This field appears only if the Meet-me Conference field is **y**. This field controls which user is allowed to change the access code for a Meet-me Conference VDN using a feature access code. This can be a local user or someone dialing in via remote access trunks.

Valid entries	Usage
extension number or blank	If an extension number is entered, only a user at that extension can change the access code for that VDN using a feature access code. If this field is blank, any station user that is assigned with console permissions can change the access code for that VDN using a feature access code.

Command reference

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Use the commands in the tables below to access each administration screen.

Brackets [ ] indicate the qualifier is optional. Single quotes (‘ ‘) indicate the text inside the quote must be entered exactly as shown or an abbreviated form of the word may be entered. MAX is the maximum number available in your system configuration. Some qualifiers aren’t available on all systems.

AAR and ARS Digit Analysis Table

Action	Object	Qualifier
change	aar analysis ars analysis	Enter digits between 0 to 9, ‘x’ or ‘X’ (dialed string) [‘part’ 1-8] [‘min’(1-MAX)]
display	aar analysis ars analysis	Enter digits between 0 to 9, ‘x’ or ‘X’ (dialed string) [‘part’ 1-8] [‘min’ (1-MAX)] [‘print’ or ‘schedule’]
list	aar analysis ars analysis	[‘start’ string] [‘count’ 1-MAX] [‘route’(1-MAX or r1-r32)], [‘part’ (1-8)], [‘node’ (1-MAX)], [‘to-node’ (1-MAX), [‘print’ or ‘schedule’]
list	aar route-chosen ars route-chosen	Enter dialed number, [‘partition’ (1-8)], [‘print’ or ‘schedule’]

AAR and ARS Digit Conversion Table

Action	Object	Qualifier
change	aar digit-conversion ars digit-conversion	Enter digits between 0 to 9 ‘x’ or ‘X’
display	aar digit-conversion ars digit-conversion	Enter digits between 0 to 9 ‘x’ or ‘X’ [‘print’ or ‘schedule’]
list	aar digit-conversion ars digit-conversion	Enter [‘start’ matching pattern] [‘count’ (1-MAX)] [‘print’ or ‘schedule’]

Abbreviated Dialing Lists

Action	Object	Qualifier
add	abbreviated-dialing enhanced	Enter form number
change	abbreviated-dialing enhanced	Enter form number
display	abbreviated-dialing enhanced	Enter form number
remove	abbreviated-dialing enhanced	Enter form number

Action	Object	Qualifier
add	abbreviated-dialing group	1-MAX (or ‘next’)
change	abbreviated-dialing group	1-MAX
display	abbreviated-dialing group	1-MAX [‘print’ or ‘schedule’]
list	abbreviated-dialing group	[xx][‘number’ x][‘to-number’ x][‘count’ n][‘print’ or ‘schedule’]
remove	abbreviated-dialing group	1-MAX

Action	Object	Qualifier
change	abbreviated-dialing personal	xxxx (extension number of assigned personal list) and List 1-3
display	abbreviated-dialing personal	xxxx (extension number of assigned personal list) and List 1-3 [‘print’ or ‘schedule’]
list	abbreviated-dialing personal	[xxxx (extension)][‘ext’ x][‘to-ext’ x][‘count’ n] [‘print’ or ‘schedule’]

Action	Object	Qualifier
add	abbreviated-dialing system	
change	abbreviated-dialing system	
display	abbreviated-dialing system	[‘print’ or ‘schedule’]
remove	abbreviated-dialing system	

Action	Object	Qualifier
add	abbreviated-dialing 7103A-buttons	—
change	abbreviated-dialing 7103A-buttons	—
display	abbreviated-dialing 7103A-buttons	[‘print’ or ‘schedule’]
remove	abbreviated-dialing 7103A-buttons	—

Access Endpoint

Action	Object	Qualifier
add	access-endpoint	xxxx (ext. or ‘next’)
change	access-endpoint	xxxx (ext. or ‘next’)
display	access-endpoint	xxxx (ext. or ‘next’) [‘print’ or ‘schedule’]
duplicate	access-endpoint	xxxx (ext. or ‘next’)
remove	access-endpoint	xxxx (ext. or ‘next’)
list	access-endpoint	xxxx [‘count’ 1-MAX] [‘print’ or ‘schedule’]

Administered Connection

Action	Object	Qualifier
add	administered-connection	1-MAX (or ‘next’)
change	administered-connection	1-MAX (or ‘next’)
display	administered-connection	1-MAX (or ‘next’) [‘print’ or ‘schedule’]
duplicate	administered-connection	1-MAX (or ‘next’)
remove	administered-connection	1-MAX (or ‘next’)
list	administered-connection	1-MAX [‘count’ 1-MAX] [‘print’ or ‘schedule’]

Administration Change Notification

Action	Object	Qualifier
notify	history	

Alias Station

Action	Object	Qualifier
change	alias station	
display	alias station	[‘print’ or ‘schedule’]

Alphanumeric Dialing Table

Action	Object	Qualifier
change	alphanumeric-dial-table	
display	alphanumeric-dial-table	[‘print’]

Announcements/Audio Sources

Action	Object	Qualifier
change	announcements	
change	integ-annc-brd-loc	
display	announcements	[‘print’ or ‘schedule’]
display	integrated-annc-boards	[‘print’ or ‘schedule’]
erase	announcements	board-location
list	integrated-annc-boards	board-location [‘print’ or ‘schedule’]
copy	announcements	
restore	announcements	
save	announcements	

ARS Toll Table

Action	Object	Qualifier
change	ars toll	n:xxx (1–MAX:office code) n:xyy: n(1-MAX); ‘:’ ;x(2-MAX;
display	ars toll	n:xxx (1–MAX:office code) [‘print’ or ‘schedule’]

ATM Board

Action	Object	Qualifier
status	atm-board	

ATM WSP

Action	Object	Qualifier
add	atm-wsp	1-15 (or ‘next’)

Attendant Console

Action	Object	Qualifier
add	attendant	1-MAX
change	attendant	1-MAX
display	attendant	1-MAX [‘print’ or ‘schedule’]
remove	attendant	1-MAX

Authorization Code — COR Mapping

Action	Object	Qualifier
change	authorization-code	auth (4-13 digit number)
display	authorization-code	auth (4-13 digit number)
list	authorization-code	none

Boot Image

Action	Object	Qualifier
get	boot-image	The physical location of the circuit pack (UUCSS)
set	boot-image	The physical location of the circuit pack (UUCSS) 1 (Directs the system to use the Image 1 firmware file) 2 (Directs the system to use the Image 2 firmware file)

Bulletin Board

Action	Object	Qualifier
change	bulletin-board	
display	bulletin-board	[print or schedule]

Button Type

Action	Object	Qualifier
list-usage	button-type	[‘print’ or ‘schedule’]

Call Vector

Action	Object	Qualifier
change	vector	1-MAX
display	vector	1-MAX [‘print’ or ‘schedule’]

CAMA Numbering Format

Action	Object	Qualifier
change	cama-numbering	—
display	cama-numbering	[‘print’ or ‘schedule’]

CDR System Parameters

Action	Object	Qualifier
change	system-parameters cdr	—
display	system-parameters cdr	[‘print’ or ‘schedule’]

Class of Restriction

Action	Object	Qualifier
change	cor	0-95
display	cor	0-95 [‘print’ or ‘schedule’]
list	cor	0-95 [‘print’ or ‘schedule’]

Class of Service

Action	Object	Qualifier
change	cos	—
display	cos	[‘print’ or ‘schedule’]

Code Calling IDs

Action	Object	Qualifier
change	paging code-calling-ids	
display	paging code-calling-ids	[‘print’ or ‘schedule’]

Command Permission Categories

Action	Object	Qualifier
change	permissions	login-id [‘print’ or ‘schedule’]
display	permissions	login-id [‘print’ or ‘schedule’]

Console-Parameters

Action	Object	Qualifier
change	console-parameters	—
display	console-parameters	[‘print’ or ‘schedule’]

Coverage Answer Group

Action	Object	Qualifier
add	coverage answer-group	1-MAX (or ‘next’)
change	coverage answer-group	1-MAX
display	coverage answer-group	1-MAX [‘print’ or ‘schedule’]
list	coverage answer-group	[1-MAX][‘number’ x][‘to-number’ x] [‘name’ x][‘count’ n][‘print’ or ‘schedule’]
remove	coverage answer-group	1-MAX

Coverage Path

Action	Object	Qualifier
add	coverage path	1-MAX (or ‘next’)
change	coverage path	1-MAX
display	coverage path	1-MAX [‘print’ or ‘schedule’]
list	coverage path	[1-MAX][‘path’ x] [‘to-path’ x][‘count’ n] [‘print’ or ‘schedule’]
remove	coverage path	1-MAX

CTI-Link

Action	Object	Qualifier
add	cti-link	link number
change	cti-link	link number
display	data-module	link number
list	cti-link	link number
list	usage cti-link	link number
remove	cti-link	link number

Data Modules

Action	Object	Qualifier
add	data-module	extension (or ‘next’)
change	data-module	extension
display	data-module	extension [print or schedule]
list	data-module	—

Date and Time

Action	Object	Qualifier
set	time	—
display	time	[‘print’ or ‘schedule’]

Daylight Savings Rule

Action	Object	Qualifier
change	daylight-savings-rules	—
display	daylight-savings-rules	[‘print’ or ‘schedule’]

Dial Plan Analysis Table

Action	Object	Qualifier
change	dialplan analysis	—
display	dialplan analysis	[‘print’ or ‘schedule’]

Dial Plan Parameters

Action	Object	Qualifier
change	dialplan parameters	—
display	dialplan parameters	[‘print’ or ‘schedule’]

Digit Absorption

Action	Object	Qualifier
change	digit-absorption	0-4
display	digit-absorption	0-4 [‘print’ or ‘schedule’]

Digit String

Action	Object	Qualifier
list-usage	digit-string	[‘print’ or ‘schedule’]

DS1 Circuit Pack

Action	Object	Qualifier
add	ds1	[P]Css
change	ds1	[P]Css(board)
display	ds1	[P]Css
list	measurements ds1 summary	location
list	measurements ds1 log	location
remove	ds1	[P]Css

Ethernet Option

Action	Object	Qualifier
get	ethernet-option	The physical location of the circuit pack (UUCSS)
set	boot-image	The physical location of the circuit pack (UUCSS)

Extended Pickup Group

Action	Object	Qualifier
add	extended-pickup -group	1-MAX (or ‘next’)
change	extended-pickup -group	1-MAX
display	extended-pickup -group	1-MAX [‘print’ or ‘schedule’]
list	extended-pickup -group	[1-MAX][‘number’ x][‘to-number’ x][‘count’ n] [‘print’ or ‘schedule’]
remove	extended-pickup -group	1-MAX

Extension

Action	Object	Qualifier
list-usage	extension	xxxx (xxxx is the path replacement extension) [‘print’ or ‘schedule’]

Extensions Administered to have an MCT-Control Button

Action	Object	Qualifier
change	mct-group-extensions	—

Extension Type

Action	Object	Qualifier
list	list-extension-type	

Feature Access Code

Action	Object	Qualifier
change	feature-access-codes	—
display	feature-access-codes	[‘print’ or ‘schedule’]

Feature-Related System Parameters

Action	Object	Qualifier
change	system-parameters features	—
display	system-parameters features	[‘print’ or ‘schedule’]

Group Paging

Action	Object	Qualifier
add	group-page	1-MAX (or ‘next’)
change	group-page	1-MAX
display	group-page	1-MAX [‘print’ or ‘schedule’]

History

Action	Object	Qualifier
list	list-history	

Holiday Table

Action	Object	Qualifier
list-usage	holiday-table	[‘print’ or ‘schedule’]

Hospitality

Action	Object	Qualifier
change	system-parameters hospitality	—
display	system-parameters hospitality	[‘print’ or ‘schedule’]

Hunt Group

Action	Object	Qualifier
add	hunt-group	1-MAX (or ‘next’)
change	hunt-group	1-MAX
display	hunt-group	1-MAX [‘print’ or ‘schedule’]
list	hunt-group	[‘print’ or ‘schedule’]
remove	hunt-group	1-MAX
list-usage	hunt-group	[‘print’ or ‘schedule’]
list	members hunt-group	(‘hunt group’ x)
list	members hunt-group	(‘loginid’ x ‘to-loginid’ x)
list	members hunt-group	(‘ext’ x ‘to-ext’ x)

Intercom Group

Action	Object	Qualifier
add	intercom-group	1-MAX (or ‘next’)
change	intercom-group	1-MAX
display	intercom-group	1-MAX [‘print’ or ‘schedule’]
list	intercom-group	[1-MAX][‘number’ x][‘to-number’ x][‘count’ n][‘print’ or ‘schedule’]
remove	intercom-group	1-MAX

Inter-Exchange Carrier (IXC) codes

Action	Object	Qualifier
change	ixc-codes	
display	ixc-codes	[‘print’ or ‘schedule’]

Intra-Switch CDR

Action	Object	Qualifier
add	intra-switch-cdr	
change	intra-switch-cdr	[extension]
display	intra-switch-cdr	[extension][‘print’ or ‘schedule’]
list	intra-switch-cdr	[extension][count x]

IP Address

Action	Object	Qualifier
list-usage	ip-address	[‘print’ or ‘schedule’]

IP Codec Set

Action	Object	Qualifier
change	ip-codec-set	Codec set number (1-7)
display	ip-codec-set	Codec set number (1-7) [‘print’ or ‘schedule’]

IP Interfaces

Action	Object	Qualifier
change	ip-interfaces	
display	ip-interfaces	[‘print’ or ‘schedule’]

IP Network Region

Action	Object	Qualifier
change	ip-network-region	Network region number (1-250)
display	ip-network-region	Network region number (1-250) [‘print’ or ‘schedule’]

IP Route

Action	Object	Qualifier
change	ip-route	IP Route number (1-400)
display	ip-route	IP Route number (1-400) [‘print’ or ‘schedule’]

IP Services

Action	Object	Qualifier
change	ip-services	
display	ip-services	[‘print’ or ‘schedule’]

ISDN Numbering — Private

Action	Object	Qualifier
change	isdn	private-numbering
display	isdn	private-numbering

ISDN Numbering — Public/ Unknown

Action	Object	Qualifier
change	isdn	public-unknown-numbering
display	isdn	public-unknown-numbering

ISDN-BRI Trunk Circuit Pack

Action	Object	Qualifier
add	bri-trunk-board	PPCSS
change	bri-trunk-board	PPCSS
display	bri-trunk-board	[‘print’ or ‘schedule’]
remove	bri-trunk-board	PPCSS

Language Translations

Action	Object	Qualifier
change	display-messages auto-wakeup-dn-dst	—
display	display-messages auto-wakeup-dn-dst	[‘print’ or ‘schedule’]
change	display-message button-labels	—
display	display-message button-labels	[‘print’ or ‘schedule’]
change	display-messages call-identifiers	—
display	display-messages call-identifiers	[‘print’ or ‘schedule’]
change	display-messages date-time	—
display	display-messages date-time	[‘print’ or ‘schedule’]
change	display-messages leave-word-calling	—
display	display-messages leave-word-calling	[‘print’ or ‘schedule’]

Action	Object	Qualifier
change	display-messages malicious-call-trace	—
display	display-messages malicious-call-trace	[‘print’ or ‘schedule’]
change	display-messages miscellaneous-features	—
display	display-messages miscellaneous-features	[‘print’ or ‘schedule’]
change	display-messages property-management	—
display	display-messages property-management	[‘print’ or ‘schedule’]
change	display-messages softkey-labels	—
display	display-messages softkey-labels	[‘print’ or ‘schedule’]
change	display-messages time-of-day-routing	—
display	display-messages time-of-day-routing	[‘print’ or ‘schedule’]
change	display-messages transfer-conference	—
display	display-messages transfer-conference	[‘print’ or ‘schedule’]

Listed Directory Numbers

Action	Object	Qualifier
change	listed-directory-numbers	—
display	listed-directory-numbers	[‘print’ or ‘schedule’]

Locations

Action	Object	Qualifier
change	locations	—
display	locations	[‘print’ or ‘schedule’]

Login Administration

Action	Object	Qualifier
add	login	login-id
change	login	login-id
display	login	login-id [‘print’ or ‘schedule’]

Loudspeaker Paging

Action	Object	Qualifier
change	paging loudspeaker	—
display	paging loudspeaker	[‘print’ or ‘schedule’]

Measurements Announcements

Action	Object	Qualifier
list	measurements-announcements-all	[‘yesterday-peak’, ‘today-peak’ or ‘last-hour’]
list	measurements-announcements-integ-all	[‘yesterday-peak’, ‘today-peak’ or ‘last-hour’]
list	measurements-announcements-board-loc	[‘yesterday-peak’, ‘today-peak’ or ‘last-hour’]

Media Gateway

Action	Object	Qualifier
add	media-gateway	1-MAX (or ‘next’)
list	media-gateway	[‘print’ or ‘schedule’]

Mode Code Related System Parameters

Action	Object	Qualifier
change	system-parameters mode-code	—
display	system-parameters mode-code	[‘print’]

Modem Pool Group

Action	Object	Qualifier
add	modem-pool num	[1-MAX]
change	modem-pool num	[1-MAX]
display	modem-pool num	[1-MAX] [‘print’ or ‘schedule’]
list	modem-pool num	[‘print’ or ‘remove’]
remove	modem-pool num	[1-MAX]

Multifrequency-Signaling-Related System Parameters

Action	Object	Qualifier
change	system-parameters multifrequency-signaling	
display	system-parameters multifrequency-signaling	[‘print’ or ‘schedule’]

Music Sources

Action	Object	Qualifier
change	music-sources	—
display	music-sources	[print or schedule]

Network Regions

Action	Object	Qualifier
change	ip-network - region	Enter the network region number (1-250 for r, 80 for si/csi/d)
display	ip-network - region	[‘print’ or ‘schedule’]

Node Name

Action	Object	Qualifier
list-usage	node-name	[‘print’ or ‘schedule’]

Packet Gateway Board

Action	Object	Qualifier
add	pgate	Enter the circuit pack cabinet, carrier, and slot
change	pgate	Enter the circuit pack cabinet, carrier, and slot
display	pgate	Enter the circuit pack cabinet, carrier, and slot
remove	pgate	Enter the circuit pack cabinet, carrier, and slot

Partition Route Table

Action	Object	Qualifier
change	partition-route-table	[‘index’ (1-2000)]
display	partition-route-table	[‘index’ (1-2000)] [‘print’ or ‘schedule’]

Pickup Group

Action	Object	Qualifier
add	pickup-group	1-MAX (or ‘next’)
change	pickup-group	1-MAX
display	pickup-group	1-MAX [‘print’ or ‘schedule’]
list	intercom-group	[1-MAX][‘number’ x][‘to-number’ x][‘count’ n] [‘print’ or ‘schedule’]
remove	pickup-group	1-MAX

PRI Endpoint

Action	Object	Qualifier
add	pri-endpoint	extension or 'next'
change	pri-endpoint	extension
display	pri-endpoint	extension ['print' or 'schedule']
list	pri-endpoint	extension ['count' X]['print' or 'schedule']
remove	pri-endpoint	extension

Remote Access

Action	Object	Qualifier
change	remote-access	—
display	remote-access	['print' or 'schedule']
list	remote-access	—
status	remote-access	['print']

Remote Call Coverage Table

Action	Object	Qualifier
change	coverage	remote
display	coverage	remote

RHNPA Table

Action	Object	Qualifier
change	rhnpa	Enter RHNPA and office code n:xyy n(1-MAX) x(0-MAX) y(0-MAX) y(0-MAX)
display	rhnpa	Enter RHNPA and office code n:xyy n(1-MAX) x(0-MAX) y(0-MAX) y(0-9) ['print' or 'schedule']

Route Pattern

Action	Object	Qualifier
change	route-pattern	1-MAX
display	route-pattern	1-MAX [print]
list	route-pattern	Enter [‘trunk’ (1-MAX)] [‘service’/feature name string] [‘print‘ or ‘schedule’]

Security-Related System Parameters

Action	Object	Qualifier
change	system-parameters security	—
display	system-parameters security	[‘print’ or ‘schedule’]

Site Data

Action	Object	Qualifier
change	site-data	—
display	site-data	[‘print’ or ‘schedule’]

Station

Action	Object	Qualifier
add	station	extension (or ‘next’)
change	station	extension
display	station	extension [‘print’ or ‘schedule’]
list	station	
list	station-sbs	
status	station	extension

System Parameters Call Coverage/Call Forwarding

Action	Object	Qualifier
change	system-parameters coverage-forwarding	—
display	system-parameters coverage-forwarding	[‘print’ or ‘schedule’]

System Parameters Country-Options

Action	Object	Qualifier
display	system-parameters country-options	[‘print’ or ‘schedule’]

System Parameters Crisis Alert

Action	Object	Qualifier
change	system-parameters crisis-alert	—
display	system-parameters crisis-alert	[‘print’ or ‘schedule’]

System Parameters Customer-Options

Action	Object	Qualifier
display	system-parameters customer-options	[‘print’ or ‘schedule’]

System Parameters IP Options

Action	Object	Qualifier
change	system-parameters ip-options	—
display	system-parameters ip-options	[‘print’ or ‘schedule’]

Telecommuting Access

Action	Object	Qualifier
add	telecommuting-access	
change	telecommuting-access	
display	telecommuting-access	x[print or schedule]
remove	telecommuting-access	

Tenant

Action	Object	Qualifier
change	tenant	x
display	tenant	x[‘print’ or ‘schedule’]

Terminal Parameters

Action	Object	Qualifier
change	terminal-parameters 6400/607A1/4600/2420	[‘print’ or ‘schedule’]
display	terminal-parameters 603/606/302B1	[‘print’ or ‘schedule’]
display	terminal-parameters 8400	[‘print’ or ‘schedule’]
display	terminal-parameters 6400	[‘print’ or ‘schedule’]

Terminating Extension Group

Action	Object	Qualifier
add	term-ext-group	1-32 (or ‘next’)
change	term-ext-group	1-32
display	term-ext-group	1-32 [‘print’ or’ schedule’]
list	term-ext-group	[1-32][‘number’ x][‘to-number’ x] [‘name’ x][‘ext’ x][‘to-ext’ x][‘count’ n] [‘print’ or ‘schedule’]
remove	term-ext-group	1-32

Terminal Parameters

Action	Object	Qualifier
change	terminal-parameters 6400/607A1/4600/2420	
display	terminal-parameters 6400/607A1/4600/2420	

TFTP Server

Action	Object	Qualifier
change	tftp-server	
display	tftp-server	

Time of Day Coverage Table

Action	Object	Qualifier
add	coverage time-of-day	1 - 999
change	coverage time-of-day	1 - 999
display	coverage time-of-day	1 - 999
remove	coverage time-of-day	1 - 999

Time of Day Routing Plan

Action	Object	Qualifier
display	time-of-day	[‘print’ or ‘schedule’]
display	time-of-day	1-MAX (plan number) [‘print’ or ‘schedule’]
change	time-of-day	[time of day routing plan (1-MAX)]

Toll Analysis

Action	Object	Qualifier
change	toll	Enter digits between 0-9, ‘x’, or ‘X’ [‘min’ 1-23]
display	toll	Enter digits between 0-9, ‘x’, or ‘X’ [‘min’ 1-23] [‘print’ or ‘schedule’]
list	toll toll-list	[‘start’ dialed-string] [‘count’ (1-1000)] [‘print’ or ‘schedule’]
list	toll all	[‘start’ dialed-string] [‘count’ (1-1000)] [‘print’ or ‘schedule’]
list	toll restricted-call	[‘start’ dialed-string] [‘count’ (1-1000)] [‘print’ or ‘schedule’]
list	toll unrestricted-call [1-10]	[‘start’ dialed-string] [‘count’ (1-1000)] [‘print’ or ‘schedule’]

Trace

Action	Object	Qualifier
list	trace-station	xxxxx/a (xxxxx is the station number and /a means the ATM specific data)
list	trace-tac	xxx/a (xxx is the trunk access code and /a means the ATM specific data

Trunk Groups

Action	Object	Qualifier
add	trunk-group	1–MAX (or ‘next’)
change	trunk-group	1–MAX or TAC X...
display	trunk-group	1–MAX [‘number’ X][‘to-number’ X][‘count’ N] [‘tac’ assigned TAC][‘print’ or ‘schedule’]
list	trunk-group	[‘print’ or ‘schedule’]
remove	trunk-group	1–MAX or TAC X...

Uniform Dial Plan

Action	Object	Qualifier
change	uniform-dialplan	Enter 1-7 digits between 0-9
display	uniform-dialplan	Enter 1-7 digits between 0-9 [‘print’ or ‘schedule’]
list	uniform-dialplan	[start *digits*] [len *length*] [insert *digits*] [net *network*] [node *node number*] [to-node *node number*] [count *number*] [‘print’ or ‘schedule’]

Vector

Action	Object	Qualifier
change	vector	link number
list-usage	vector	[‘print’ or ‘schedule’]

Vector Directory Numbers

Action	Object	Qualifier
add	vdn	extension (or ‘next’)
change	vdn	extension
display	vdn	extension [‘print’ or ‘schedule’]

Phone reference

20

This reference section describes many of the telephones and adjuncts that you can connect to Avaya MultiVantage systems.

Use this section to:

- determine where to connect a phone—is it analog, digital, hybrid, or IP? Is it designed for a 2-wire or 4-wire environment?
- determine whether a phone will fit your users' needs—does it accommodate enough feature buttons? does it include a display?
- understand how to assign the user's feature buttons—how do the buttons on the Station screen map to buttons on the physical phone.

Because you need to know how the physical buttons on the phone relate to your button assignments on the Station screen, this section includes figures for those phones to which you can assign feature buttons.

This section includes descriptions of the following telephones:

- 731x-series hybrid telephones (page 1289)
- 7400-series telephones (page 1293)
- ISDN telephones (7500-series & 8500-series) (page 1310)
- 8110 telephones (page 1315)
- 8400-series telephones (page 1315)
- CALLMASTER telephones (page 1322)
- Cordless telephones (page 1326)
- Internet Protocol (IP) SoftPhones (page 1329)

500 telephones

The 500 telephones are single appearance analog rotary-dial telephones which provides cost-effective service wherever it is located. It provides limited access to features because the rotary dial has no * or # positions.

2420 telephones

The 2420 telephone is a digital phone with an optional feature expansion module and downloadable call appearance/feature buttons information. The 2420 DCP phone does not need paper labels. The button information appears on a screen on the phone. The firmware for the 2420 can be changed via TFTP over a TCP/IP connection.

Avaya recommends that you administer call appears only on the first 8 call appearance/feature buttons on the 2420 phone.

2500-series telephones

The 2500-series telephones consist of single appearance analog telephones with conventional touch-tone dialing. You can allow 2500-series phones users to access features by giving them the appropriate feature access codes. For more information about providing feature access codes to your users, refer to ‘‘Changing feature access codes’’ on page 68.

4600-series IP telephones

The 4600-series telephones are DCP telephones that use Internet Protocol (IP) technology with Ethernet line interfaces. Firmware for these phones can be downloaded from the internet. The 4600 series includes a 2-button set, 6-button sets, a 12-button set, and a 24-button set.

Each of the 4600-series phones includes 6 standard feature buttons:

- SPEAKER button, which can access a 2-way speakerphone or allow group listen
- MUTE button, which mutes the handset or speakerphone microphone
- HOLD button
- REDIAL button
- TRANSFER/TEST button for transferring a call or testing the lights and display on the telephone
- CONF/RING button for setting up a conference call and for selecting a personalized ringing pattern.

The 4604, 4612, and 4624 phones do not have a standard Drop button, but you can assign a drop button to any feature button. The 4600-series display phones show the date and time in Normal mode, so you do not have to assign a Date/Time button to these phones.

4602 IP telephone

The 4602 IP telephone has separate LEDs to indicate the on/off status of the speakerphone and mute buttons. The 4602 has a 2x24 display. The 4602 has no administrable feature buttons, 2 fixed call appearance buttons, one-way speakerphone or no speakerphone option, a fixed Drop button, a listen-only Speaker button, a redial button, and a fixed voice mail retrieval button that provides automatic access to voice mail.

4606 IP telephones

The 4606 IP telephone is a multi-appearance digital telephone with six call appearance/feature buttons: a red Hold button, a Redial button, a Transfer button that can also be used for the Test feature, and a Conference button that can also be used to select a personalized ringing pattern, a red Message light, and a Volume control button.

4612 IP telephone

The 4612 IP telephone is a multi-appearance digital telephone with 12 call appearance/feature buttons. With the 4612 IP telephone the end-user can access 12 features with the softkeys and display control buttons.

4620 IP telephone

The 4620 IP telephone has an optional feature expansion module and downloadable call appearance/feature button information. It also includes features such as speed dial, call log, and Web browsing using the Wireless Markup Language (WML). The 4620 IP phone does not need paper labels. The button information appears on a screen on the phone.

The 4620 uses icons to indicate the status of call appearances, bridge call appearances and features. The phone maintains a call log with calling party and called party information. The 4620 has a local button for headset on/off. The button label information for the 4620 is downloaded automatically to the phone when a link is established between the switch and the phone. There are three speakerphone options on the 4620. Labels on the 4620 can be downloaded in English, French, Italian, Spanish, and user-defined languages.

4624 IP telephone

The 4624 IP telephone is a multi-appearance digital telephone with 24 call appearance/feature buttons. With the 4624 telephone the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

4630 IP screenphone

The 4630 IP Screenphone uses a single connection. A large, color, touch-sensitive screen is used to operate the telephone functions. Avaya recommends that you use the default settings on the 4630 IP Screenphone administrable feature buttons with status indication.

Figure 18. The 4630 IP Screenphone

6200-series telephones

6210 telephone

The 6210 telephones are single-line analog models. They have fixed Flash, Redial, and Hold feature buttons and a message waiting light.

6218 telephone

The 6218 telephones are single-line analog models. They have 10 programmable dialing buttons. These phones also have fixed Flash, Redial, and Hold feature buttons and a message waiting light.

6220 telephone

The 6220 telephones are single-line analog models. These phones also have fixed Flash, Redial, Hold, Mute, and Speakerphone (Spkr) feature buttons and a message waiting light. They also have handset volume control, ringer volume control, timed switch-hook disconnect, 10 programmable dialing buttons, repertory keylock, set personalized ring, and system hold.

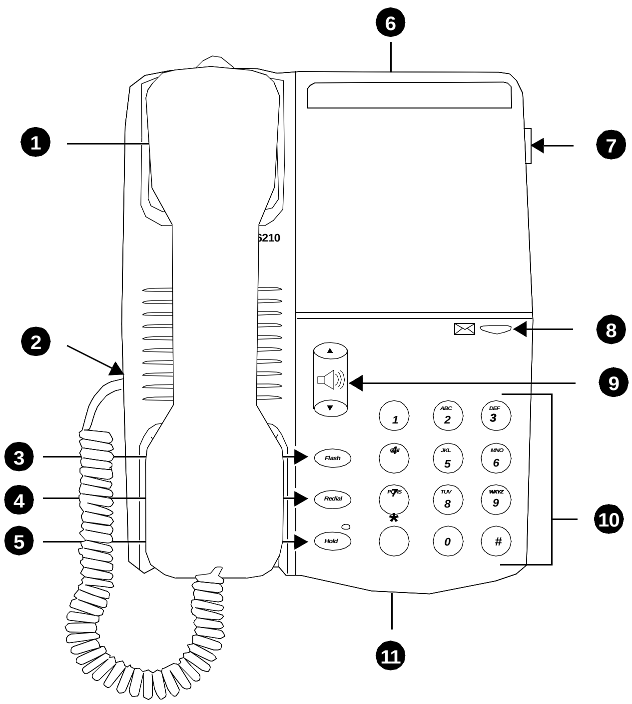


Figure Notes

1. Handset
2. Handset cord jack
3. Flash button
4. Redial button
5. Hold button
6. DATA jack
7. Ringer volume control
8. Message light
9. Handset volume control
10. Dial pad
11. LINE jack (on bottom of phone)

Figure 19. The 6210 telephone

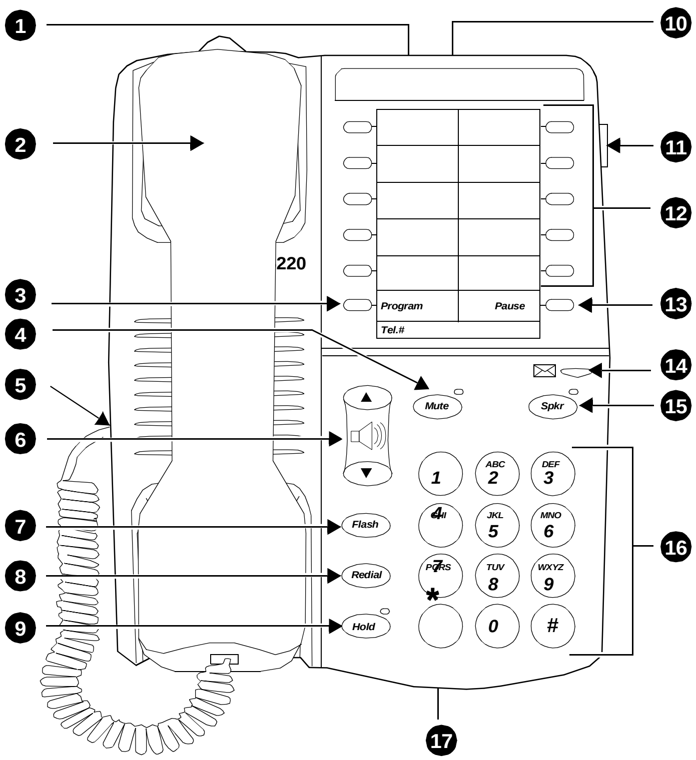


Figure Notes

1. Handset parking tab
2. Handset
3. Program button
4. Handset cord jack
5. Flash button
6. Redial button
7. Hold button
8. Data jack
9. Personalized ring
10. Ringer volume control
11. 10 programmable dialing buttons
12. Pause button
13. Message light
14. Handset/speakerphone volume control
15. Dial pad
16. LINE jack (on bottom of phone)

Figure 20. The 6218 telephone

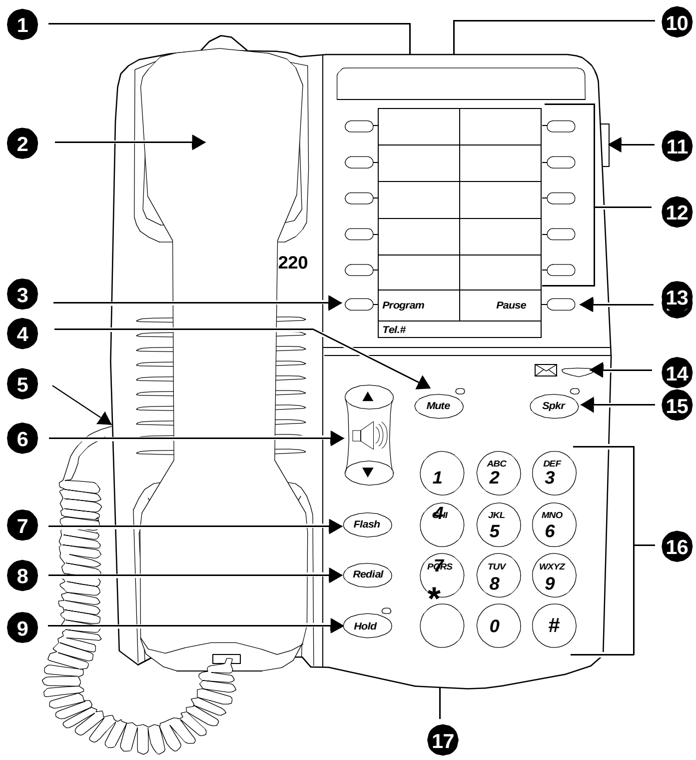


Figure Notes

1. DATA jack
2. Handset
3. Program button
4. Mute button
5. Handset cord jack
6. Handset/speakerphone volume control
7. Flash button
8. Redial button
9. Hold button
10. Personalized ring
11. Ringer volume control
12. 10 programmable dialing buttons
13. Pause button
14. Message light
15. Speakerphone button
16. Dial pad
17. LINE jack (on bottom of phone)

Figure 21. The 6220 telephone

6400-series telephones

The 6400-series telephones are DCP 2-wire telephones that work with Avaya MultiVantage. The last two digits of the 6400-series model number identify the number of call appearances (2-lamp buttons) for that model. For example, the 6424D has 24 call appearances. The 6400 series includes two single-line sets (6402 and 6402D), 8-button sets, a 16-button set, a 24-button set, and a 24-button expansion module (for the 6416D+ and 6424D+ telephones).

Each of the 6400-series phones includes 6 standard feature buttons:

- SPEAKER button, which can access a 2-way speakerphone or allow group listen
- MUTE button, which mutes the handset or speakerphone microphone
- HOLD button
- REDIAL button
- TRANSFER/TEST button for transferring a call or testing the lights and display on the telephone
- CONF/RING button for setting up a conference call and for selecting a personalized ringing pattern.

These phones do not have a standard Drop button, but you can assign a drop button to any feature button. The 6400-series display phones show the date and time in Normal mode, so you do not have to assign a Date/Time button to these phones.

6402 telephones

The 6402 is a single-line telephone with six fixed feature buttons: a listen-only Speaker button, a Feature button that allows you to use the dial pad keys for up to 12 features assigned by the system manager, a red Hold button, a Redial button, a Transfer button that can also be used for the Test feature, and a Conference button that can also be used to select a personalized ringing pattern, a red Message light, and a Volume control button.

The 6402D is the same as a 6402, but also has a 2-line by 16-character LCD display.

The 6402 and 6402D can be used in only a 2-wire environment.

6408 telephones

The 6408 is a multi-appearance digital telephone with eight call appearance/feature buttons.

The 6408 telephone is available in the following four models:

- 6408 telephone—includes a 1-way, listen-only speaker, and no display
- 6408+ telephone—includes a 2-way speakerphone, and no display
- 6408D telephone—includes a 1-way, listen-only speaker, and a 2-line by 24-character display
- 6408D+ telephone—includes a 2-way speakerphone, and a 2-line by 24-character display.

With the 6408D and 6408D+ telephones, the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

The 6408, 6408+, 6408D, and 6408D+ telephones can work only in 2-wire environments.

6416D+ telephone

The 6416D+ telephone is a multi-appearance digital telephone with 16 call appearance/feature buttons.

With the 6416D+ telephone the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

NOTE:
You can connect an XM24 expansion module to the 6416D+ telephone to expand the number of buttons you can assign. However, when the expansion module is connected, you must connect an auxiliary power supply to the telephone.

6424D+ telephone

The 6424D+ telephone is a multi-appearance digital telephone with 24 call appearance/feature buttons. With the 6424D+ telephone the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

The 6424D+ telephone can work in both 4-wire and 2-wire environments.

NOTE:
You can connect an XM24 expansion module to a 6424D+ phone to expand the number of buttons you can assign. However, when the expansion module is connected, you must connect an auxiliary power supply to the telephone.

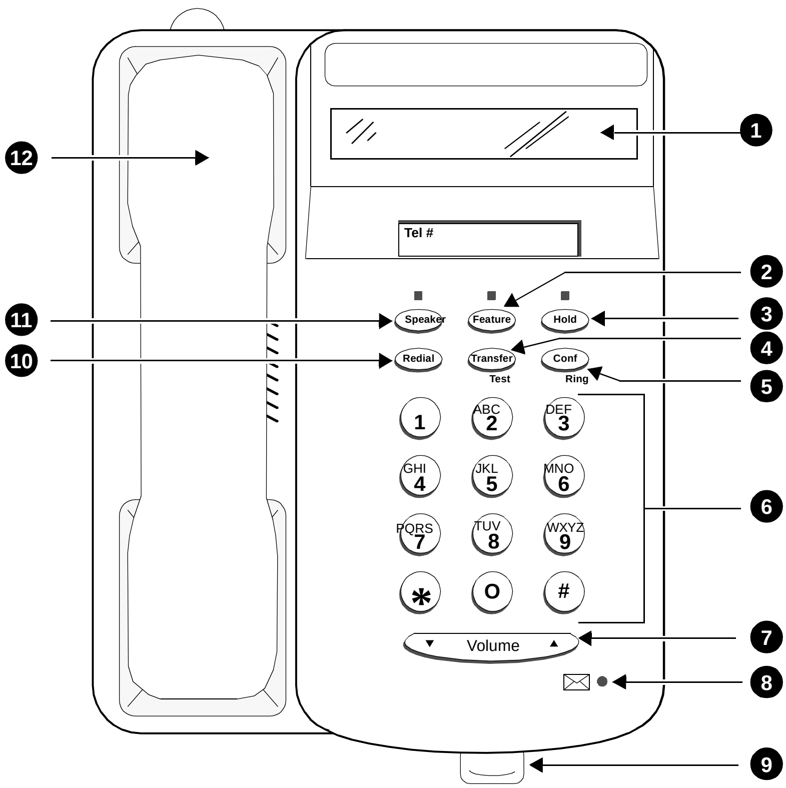


Figure Notes

1. Display
2. Feature button
3. Hold button
4. Transfer/Test button
5. Conf/Ring button
6. Dial pad
7. Volume control button
8. Message light
9. Tray handle (includes reference cards)
10. Redial button
11. Speaker button
12. Handset

Figure 22. 6402D telephone

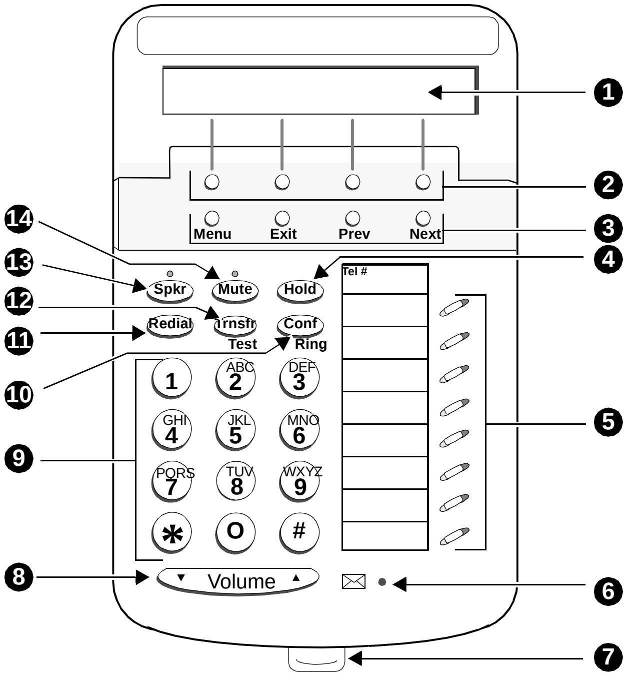


Figure Notes

1. Display
2. Softkeys
3. Display control buttons
4. Hold
5. Call appearance/feature buttons
6. Message light
7. Tray handle (includes reference cards)
8. Volume control button
9. Dial pad
10. Conf/Ring button
11. Redial button
12. Transfer/Test button
13. Speaker button
14. Mute button

Figure 23. 6408D telephone

7100-series telephones

7101A telephone

The 7101A telephone is a single-line analog model that is equipped with a Message Waiting light and a handy Recall button for activating the system's special features. It cannot be physically bridged to the same analog line port due to the message waiting and loop current circuitry.

7102A telephone

The 7102A (7102A01A) and 7102+ (7102A01B), called the 7102 Plus. The front of the two sets is exactly the same in appearance. The only difference is that the 7102A01B is equipped with an adjunct jack. This jack allows speakerphone/headset capability.

7103A telephone

The 7103A fixed feature phone is a single-line analog model which has been discontinued. The feature buttons on this phone must be programmed by the system manager. It cannot be physically bridged to the same analog line port due to the message waiting and loop current circuitry.

7104A telephone

The 7104A telephone is a single-line analog model that is equipped with a display that is used to display stored numbers. It cannot be physically bridged to the same analog line port due to the message waiting and loop current circuitry.

7300-series telephones

7302H MERLIN telephone

The 7302H is a 5-button telephone that can be desk or wall mounted. This set can no longer be ordered.

Administer 7302H telephones (5-Button) as a 7303S model.

7303H MERLIN telephone

The 7303H is a 10-button telephone that can be desk or wall mounted.

Administer 7303H telephones (10-Button) as a 7305S model. You can administer only the first 12 feature function buttons. Of these 12 buttons, 8 have two lamps and 4 have no lamps. However, the system treats the 8 double-lamp buttons as though they have a single (green) lamp. Administer the 4 buttons (with no lamps) with features that do not require status indications.

7303S telephone

The 7303S is a multi-appearance hybrid telephone which provides access to 10 line appearances or selected programmable features. The 7303S telephone is also equipped with six fixed feature buttons. It requires 3-pair wiring for operation. One wire pair is used for analog voice, while the other two pairs are used for digital control and signaling.

7305H MERLIN telephone

The 7305H series is a 34-button telephone. This telephone can be equipped with or without different features such as built-in speakerphone or display. Some versions of the 34-button series can be ordered using PEC code 3162 and the appropriate suffix.

Administer 7305H telephones (34-Button) as a 7305S model. The system treats the telephone’s 24 feature function buttons (two lamps each) as single (green) lamp function buttons.

7305S telephone

The 7305S telephone is a multi-appearance hybrid telephone which provides access to 10 line appearances. The 10 line appearance buttons can also be used as programmable feature buttons. The 7305S telephone is also equipped with 24 programmable feature buttons and six fixed feature buttons. It requires 3-pair wiring for operation. One wire pair is used for analog voice, while the other two pairs are used for digital control and signaling.

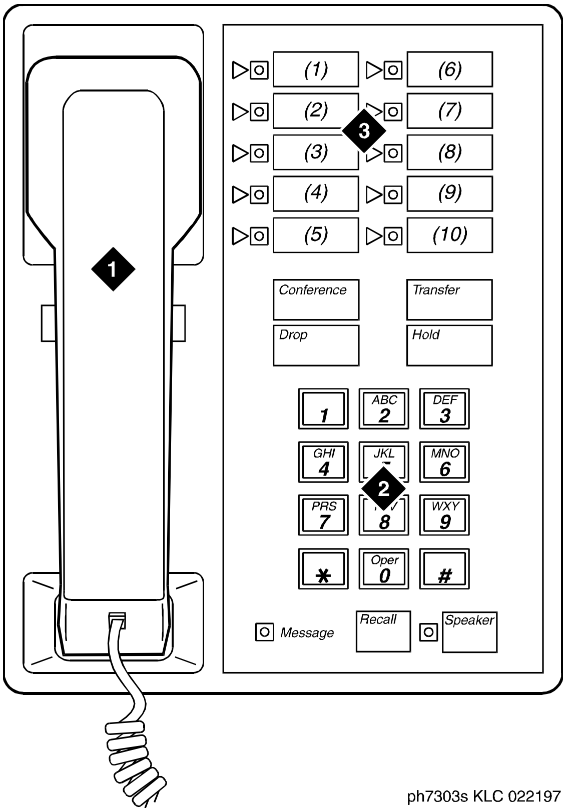


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons

Figure 24. 7303S telephone

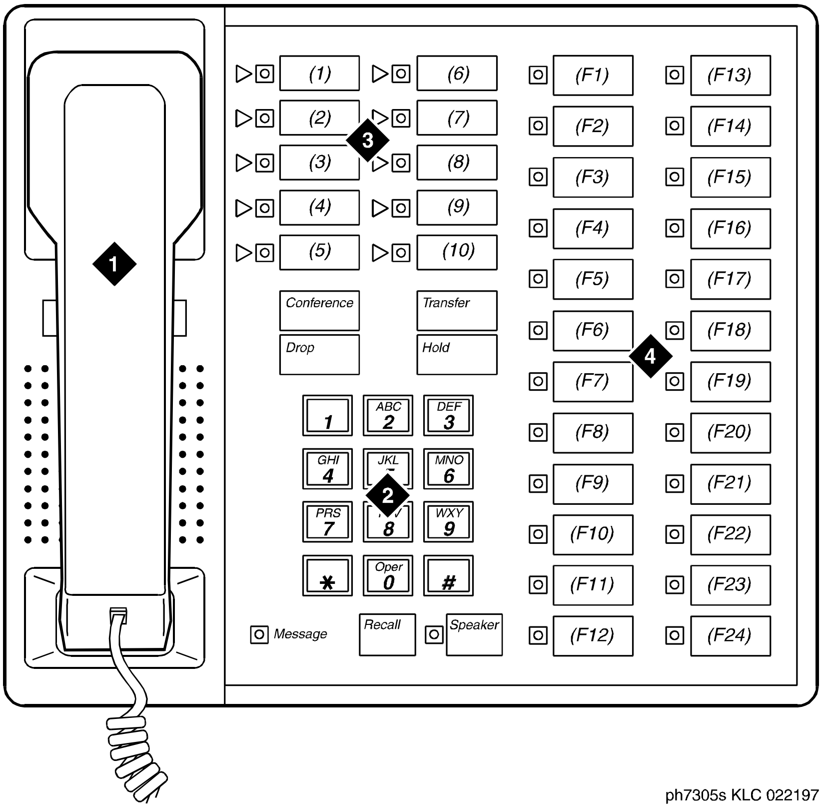


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 24 feature buttons

Figure 25. 7305S telephone

731x-series hybrid telephones

You should note that the following restrictions apply to administering hybrid telephones in the system:

- Hybrid telephones equipped with displays cannot be used as ACD agents and do not allow your end-users to access the Directory.
- The following fixed feature buttons do not operate on Hybrid telephones: STOP, PAUSE, RECALL, MESSAGE, HFAI, and HFAI/MIC. If you want users to have Hands Free Automatic Answer on Intercom (HFAI), assign Internal Automatic Answer (IAA) to a lamp button.
- These telephones support Leave Word Calling (LWC), but users cannot retrieve messages with the display.

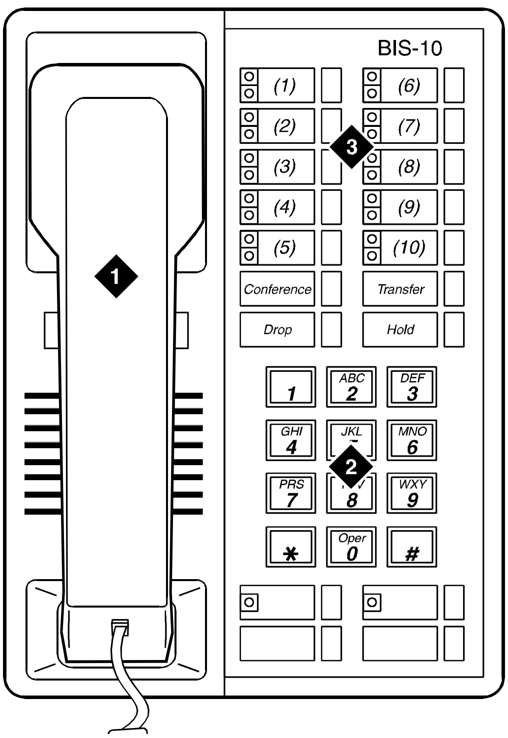


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons

Figure 26. 7313H telephone (BIS 10)

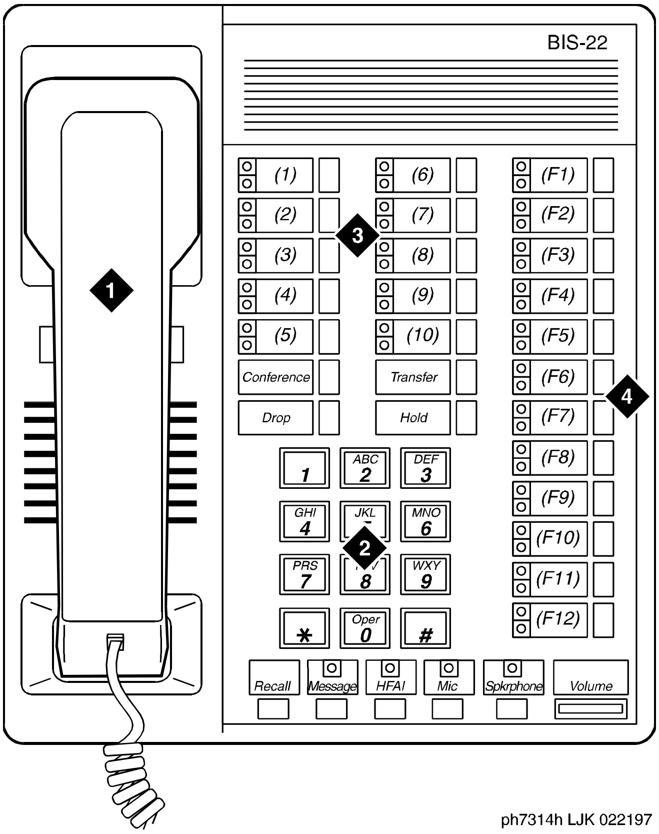


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 12 feature buttons

Figure 27. 7314H telephone (BIS 22)

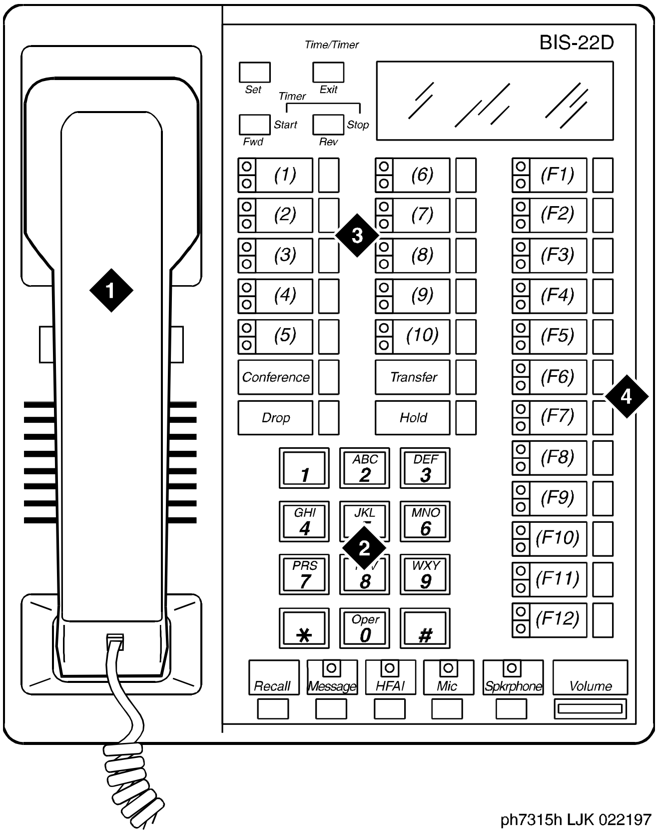


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 12 feature buttons

Figure 28. 7315H telephone (BIS 22D)

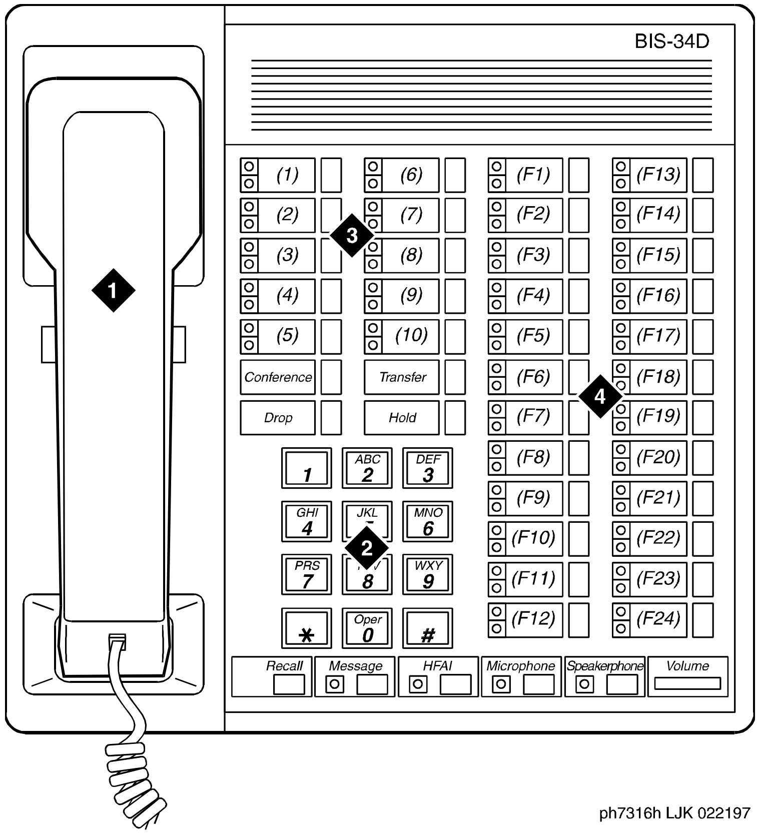


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 24 feature buttons

Figure 29. 7316H telephone (BIS 34)

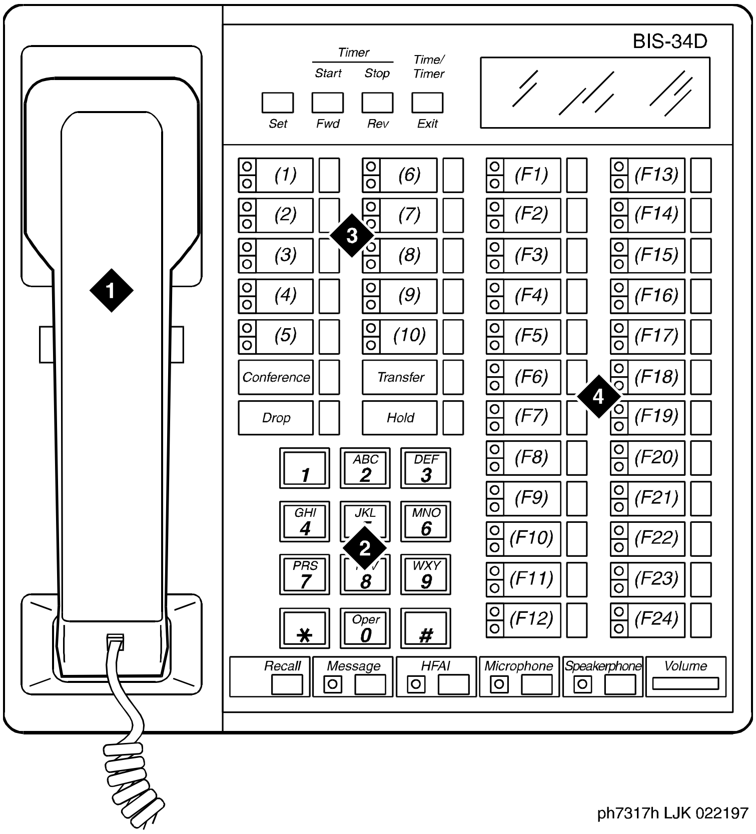


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 24 feature buttons

Figure 30. 7317H telephone (BIS 34D)

7400-series telephones

7401D telephone

The 7401D (7401D01A) and the 7401+ (7401D02A) are both single appearance digital phones which have no call appearance buttons or lights, but have two virtual call appearances. Depending on how the 7401D or the 7401+ telephone is administered, the second call appearance may be restricted to incoming priority calls and outgoing calls only.

7403D telephone

The 7403D is a multi-appearance digital telephone which has 10 buttons available for line appearances, one-touch feature access, or Speed Dialing. In addition, the 7403D telephone may be equipped with a Digital Terminal Data Module (DTDM) which attaches to the right side and allows the connection of a EIA RS-232C data terminal.

7404D telephone

The 7404D is a multi-appearance digital telephone which provides simultaneous voice/asynchronous data transmission. All the transmission is done over the same two pairs of wire.

7405D telephone

The 7405D telephone is a multi-appearance digital telephone which allows features to be added as the user needs them. The Digital Display can be added to provide access to the Message Center. A Digital Terminal Data Module or 7400B can be added to enable the user of a 7405D telephone to transmit or receive data with an associated data terminal.

The basic 7405D provides 10 call appearance/feature buttons with lights that can be assigned to call appearances or system features. It has 24 programmable feature buttons and six fixed-feature buttons. The 7405D can also have a function key module which adds 24 feature buttons and a call coverage module (when no display module is used) which adds 20 call appearance/feature buttons.

7406 telephones

The 7406D telephone (7406D01A, 7406D02A, 7406D03A, and 7406D04A models) has five call appearance/feature buttons, each with a red in-use light and a green status light, seven shiftable (2-level) programmable feature buttons with no lights, four programmable feature buttons with a green light, four fixed feature buttons (CONFERENCE, TRANSFER, DROP, and HOLD), a SHIFT button with a green light, a SPEAKER button, and a green Message light.

The 7406BIS telephone (7406D05A and 7406D06A models) has five call appearance/feature buttons, each with a red in-use light and a green status light, seven shiftable (2-level) programmable buttons with no lights, two programmable feature buttons with a green light, four fixed feature buttons (CONFERENCE, TRANSFER, DROP, and HOLD), a SHIFT button with a green light, a SPEAKER button with a green light, a MUTE button with a red light, a SPEAKER VOLUME “arrow” button, and a red Message light.

The 7406+ telephone (7406D07A and 7406D08A models) has five call appearance/feature buttons, each with a red in-use light and a green status light, three shiftable (2-level) programmable feature buttons with a green light, six shiftable (2-level) programmable feature buttons without lights, four fixed feature buttons (CONFERENCE, TRANSFER, DROP, and HOLD), a SELECT button with a green light, a SPEAKER/RESET SPKR button with a green light, a MUTE button with a red light, a VOLUME “arrow” button, and a red Message light.

7407+ telephone

The 7407D, Enhanced 7407D, and 7407+ telephones are multi-appearance digital telephones which provide digital voice, display, and data capabilities (the latter with the 7400B+ Data Module).

There are three versions of the 7407D telephone:

- 7407+ (7407D02D) — offers 10 call appearance buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with one light each (SELECT, SPEAKER/RESET SPKR, and MUTE), nine feature buttons with one light each (the uppermost two buttons can be used for voice or display features, the lower seven buttons for display features), 22 flexible feature buttons with no lights, a Message light, personalized ringing, a built-in speakerphone with a reset and listen-only option, and a built-in 2-line by 40-character display.
- The 7407D (the 7407D01B)—offers 10 call appearance/feature buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with an associated light (CALCULATOR/SELECT RING, SPEAKERPHONE, and MICROPHONE), nine programmable feature buttons with lights (the two uppermost buttons can be used for voice or display features, the lower seven for display features), 11 dual-function buttons, 22 programmable feature-only buttons without lights, a Message light, a DISPLAY button that turns the display on and off, personalized ringing, a built-in speakerphone, a 2-line by 40-character liquid crystal display, and a built-in calculator. This set is AC powered.
- Enhanced 7407D (the 7407D02C)—offers 10 call appearance buttons, each with a red in-use light and green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with an associated light (SELECT RING, SPEAKER, and MUTE), 22 programmable feature-only buttons without lights, nine programmable feature buttons with one light each (the uppermost two buttons can be used for voice or display features, the lower seven for display features), a Message light, personalized ringing, a built-in

speakerphone, a connection for an adjunct speakerphone or headset, a speakerphone with spokesman, and Mute option, and a 2-line by 40-character display.

7410D and 7410+ telephones

The 7410D (7410D01A) and 7410+ (7410D02A) are both multi-appearance digital telephones with 10 line appearances, four standard fixed feature buttons and a Select Ring button. The 7410D and 7410+ telephones also provide a Message light, Ringer Volume control, a Speakerphone/headset adapter jack. You can activate a Self-Test feature to test the lights and tone ringer on the telephone. In addition, the 7410+ provides a Speaker for listening-only functions.

7434D telephone

The 7434D is a multi-appearance digital telephone that offers 34 call appearance/feature buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with one light each (SELECT, SPEAKER/RESET SPKR, and MUTE), seven display feature buttons with one light each, a Message light, personalized ringing, a built-in speakerphone with a reset option, and a built-in 2-line by 40-character display. You can connect this phone to a digital line port. The 7434D telephone supports an adjunct display module or a call coverage module.

7444D telephone

The 7444 telephone is a multi-appearance digital telephone that offers 34 call appearance/feature buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with one light each (SELECT, SPEAKER/RESET SPKR, and MUTE), seven display feature buttons with one light each, a Message light, personalized ringing, a built-in speakerphone with a reset option, and a built-in 2-line by 40-character display. You can connect this telephone to a digital line port. It is powered from the switch.

NOTE:
The 7444 is powered by the switch, however, to use the display, you must connect an auxiliary power supply to the telephone.

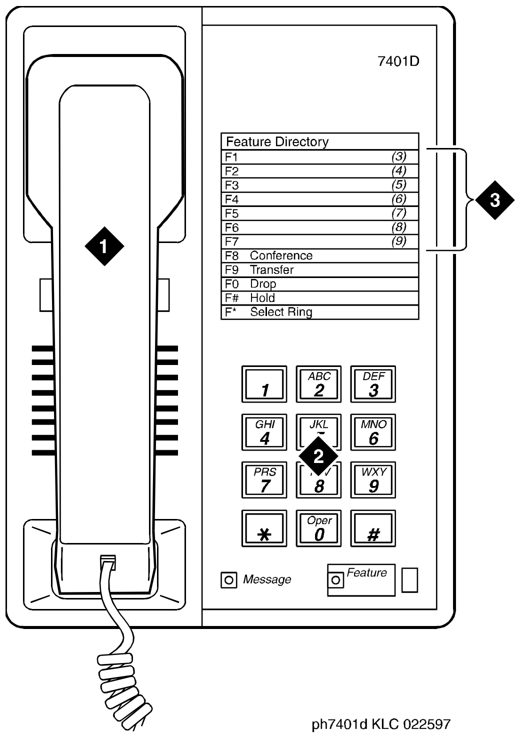


Figure Notes

1. Handset
2. Dial pad
3. Access codes card

Figure 31. 7401D telephone

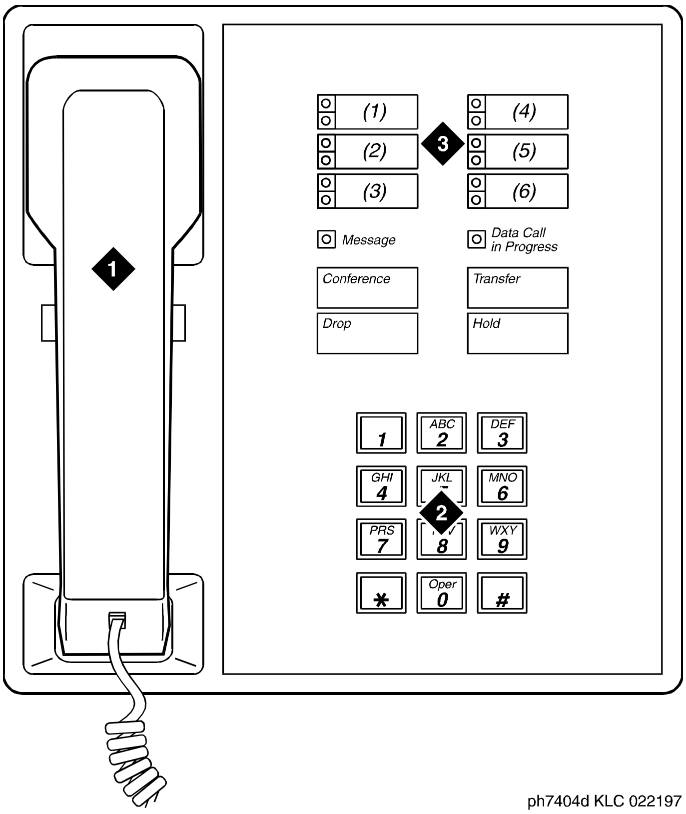


Figure Notes

1. Handset
2. Dial pad
3. 6 programmable buttons

Figure 32. 7404D telephone

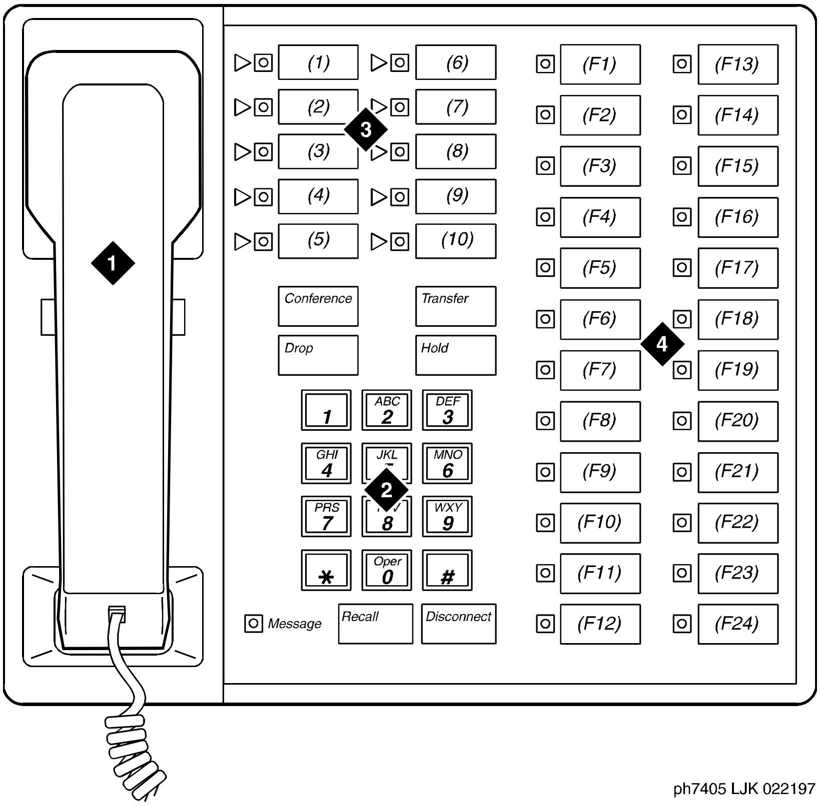


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 24 feature buttons

Figure 33. 7405D telephone

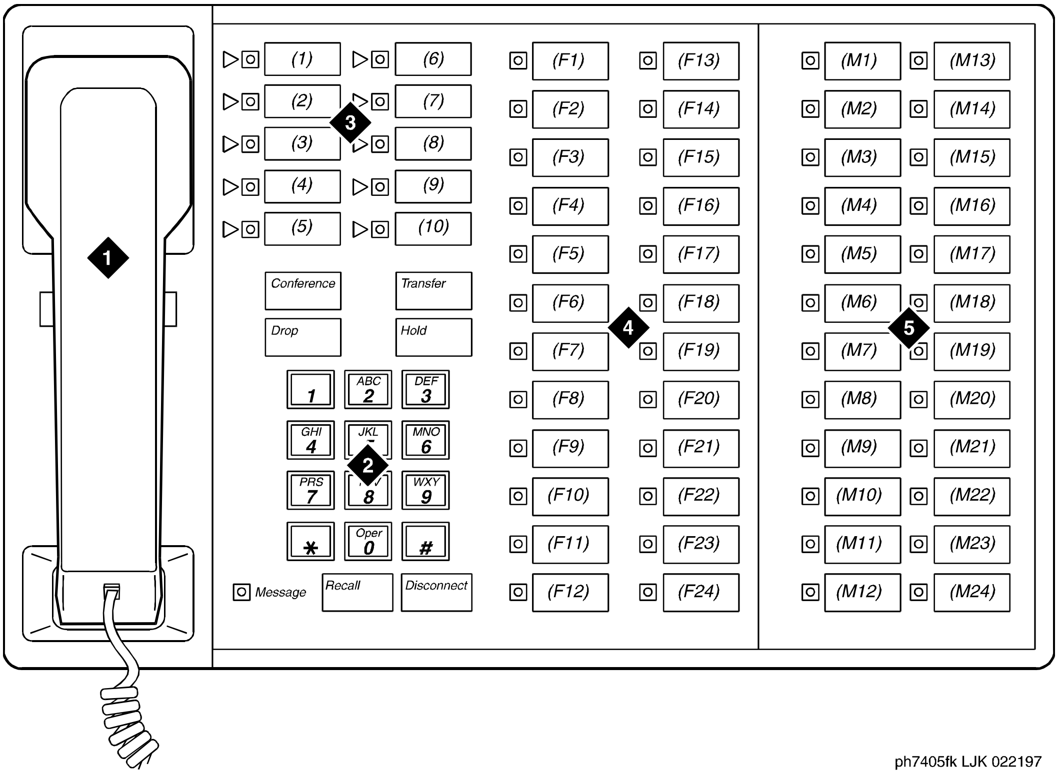


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 24 feature buttons
5. Function key module with 24 feature module buttons

Figure 34. 7405D telephone with optional function key module

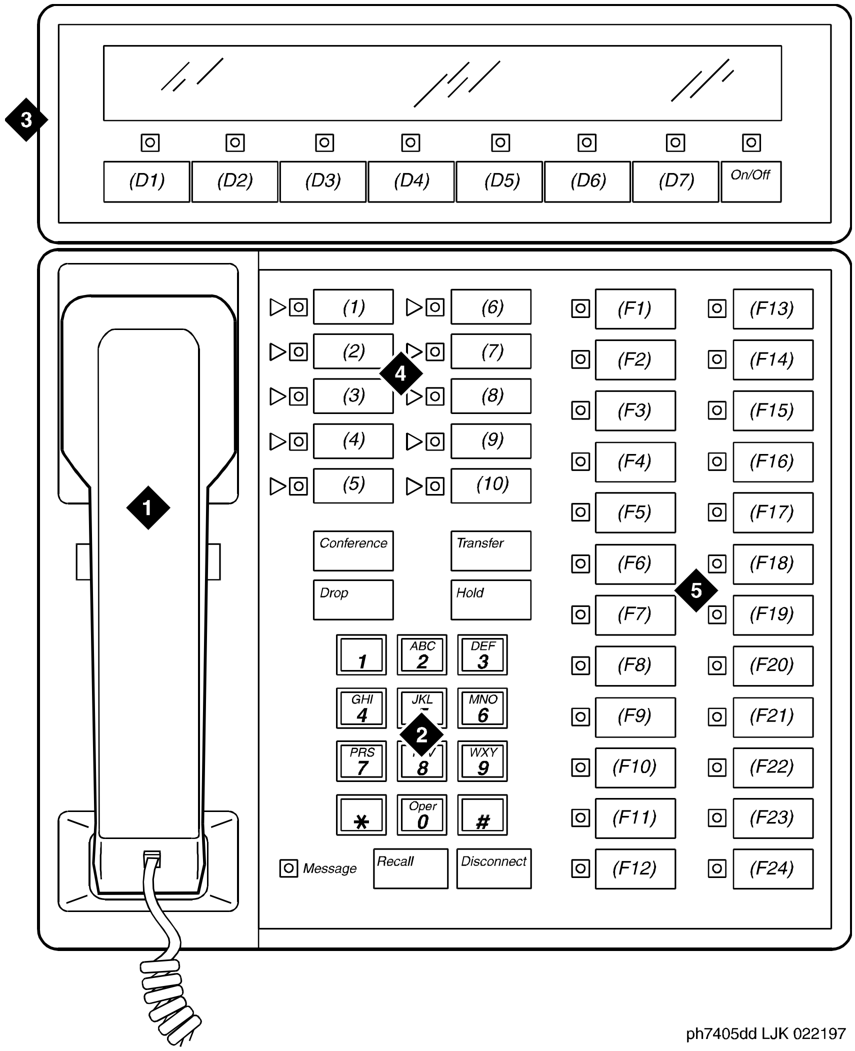


Figure Notes

1. Handset
2. Dial pad
3. Digital display module with 7 display buttons
4. 10 programmable buttons
5. 24 feature buttons

Figure 35. 7405D telephone with optional digital display module

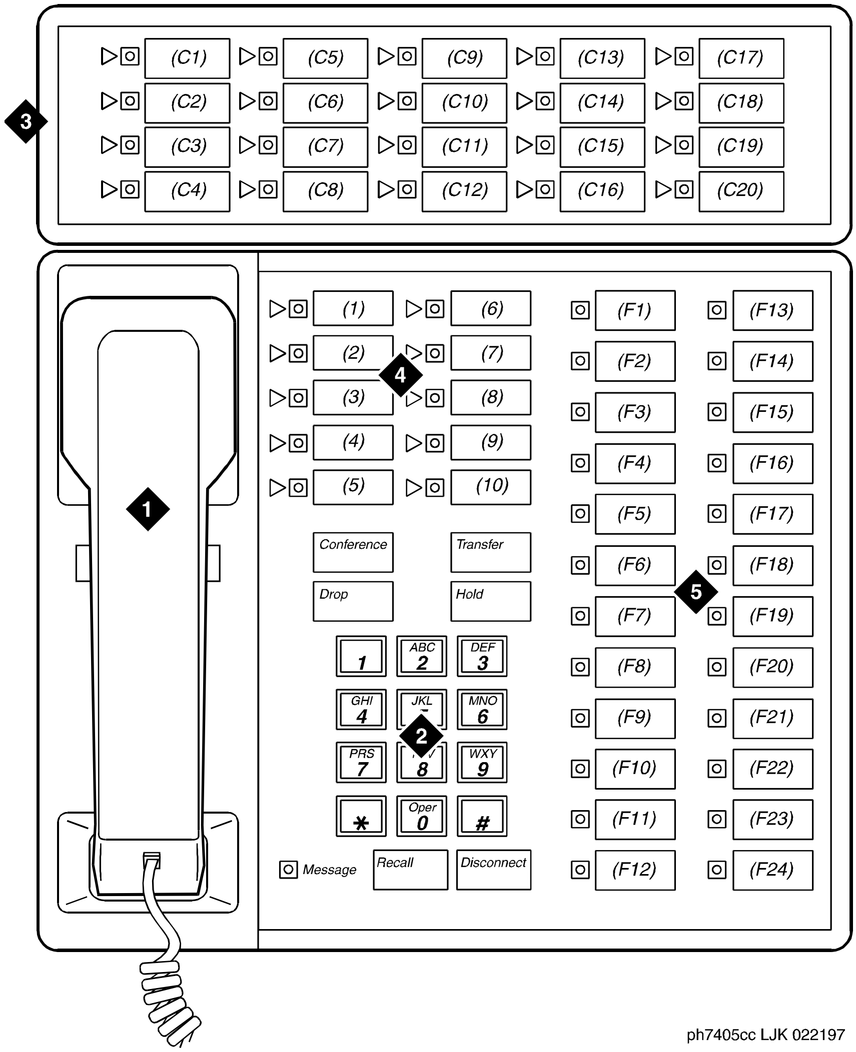


Figure Notes

1. Handset
2. Dial pad
3. Call coverage module with 20 coverage module buttons and status lamps
4. 10 programmable buttons
5. 24 feature buttons

Figure 36. 7405D telephone with optional call coverage module

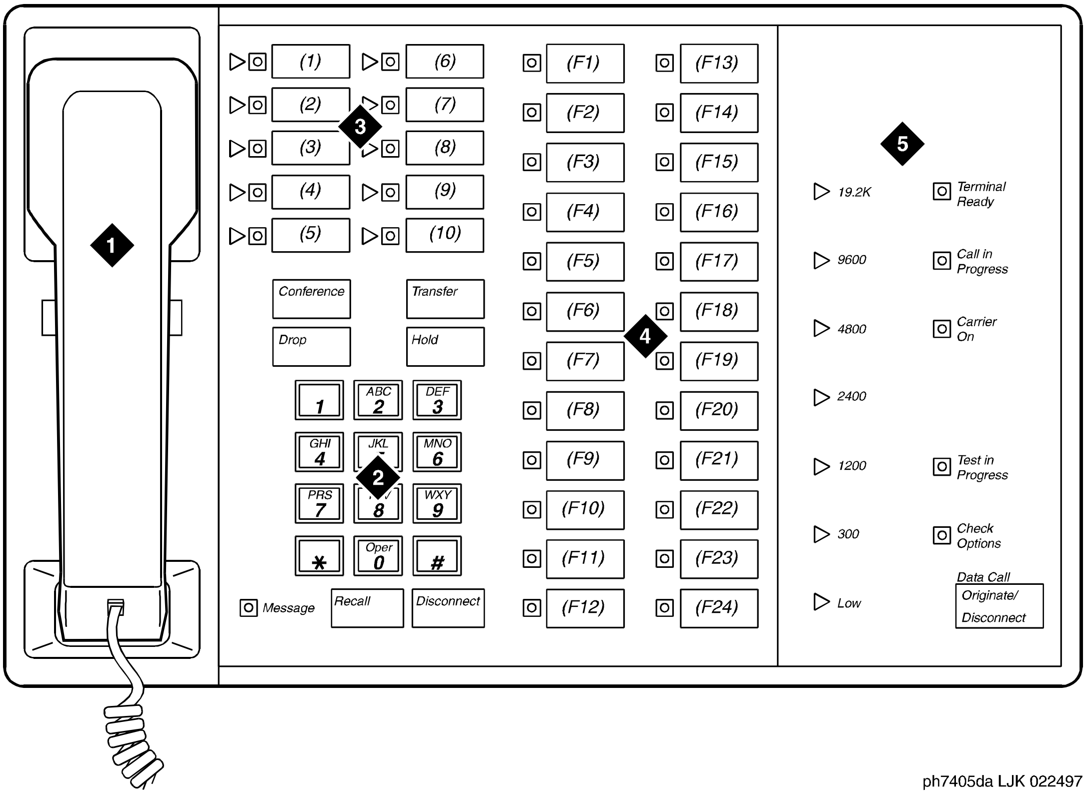


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 24 feature buttons
5. Digital terminal data module

Figure 37. 7405D telephone with optional digital terminal data module

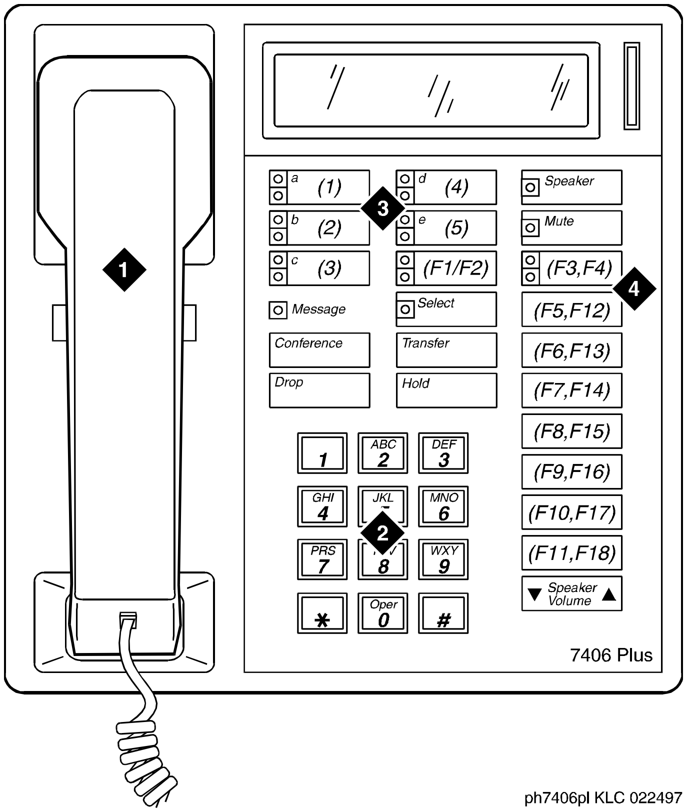


Figure Notes

1. Handset
2. Dial pad
3. 5 programmable buttons
4. 18 feature buttons (feature buttons F2, F4, and F12 to F18 are enabled with the Shift key)

Figure 38. 7406D+ telephone

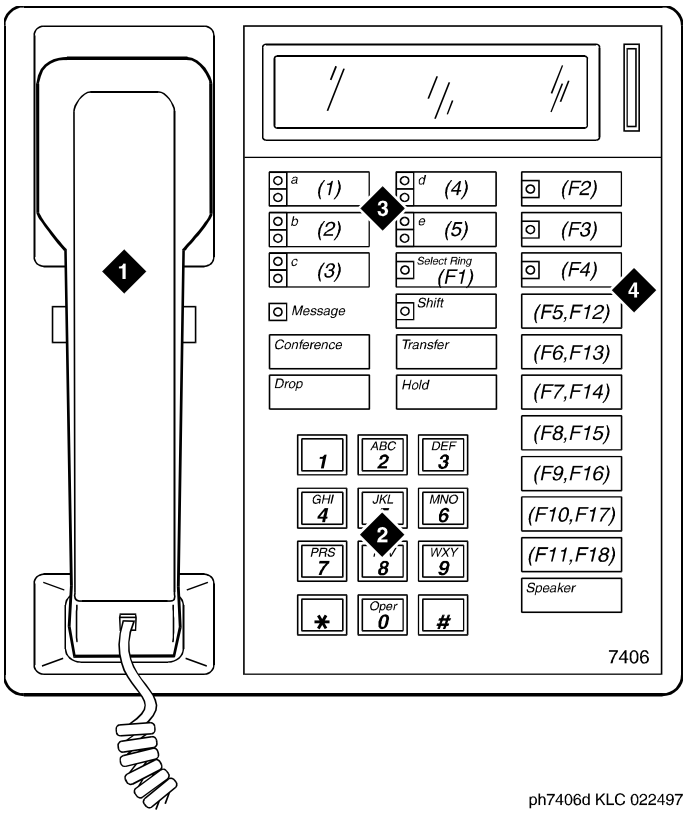


Figure Notes

1. Handset
2. Dial pad
3. 5 programmable buttons
4. 18 feature buttons (feature buttons F12 to F18 are enabled with the Shift key)

Figure 39. 7406D telephone

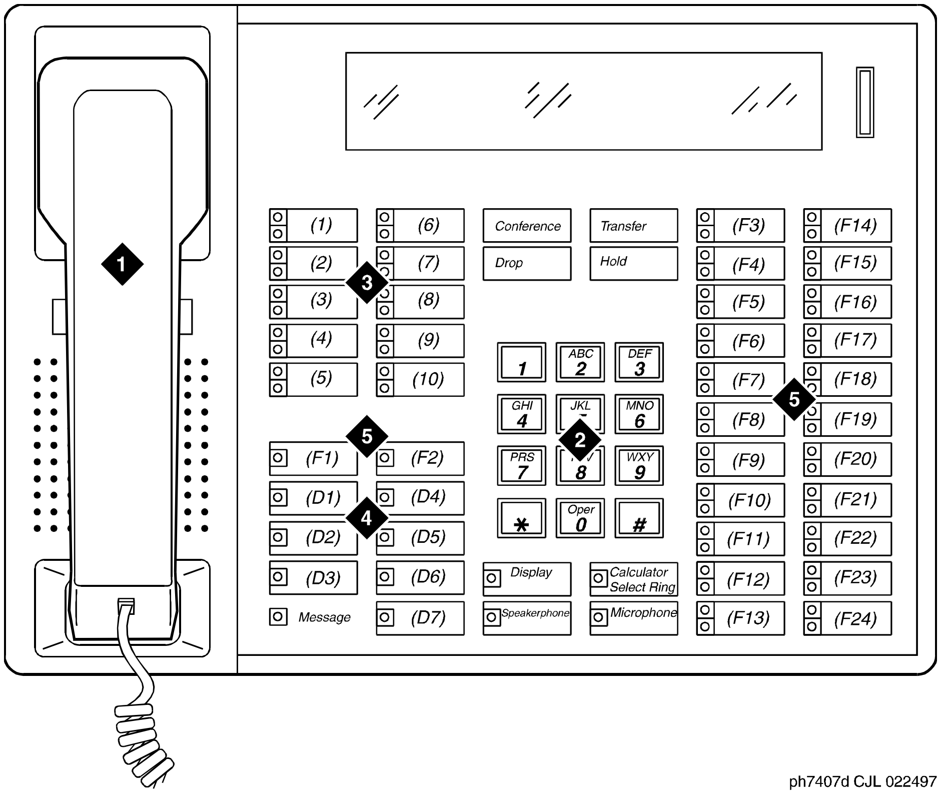


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 7 display buttons
5. 24 feature buttons

Figure 40. 7407D telephone

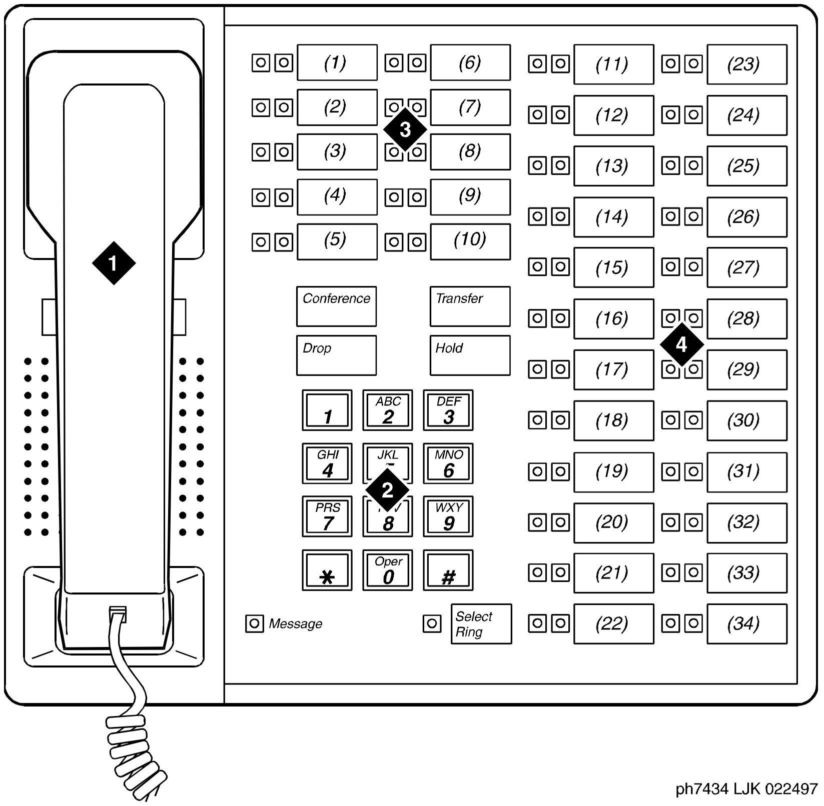


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 24 feature buttons (11 to 34)

Figure 41. 7434D telephone

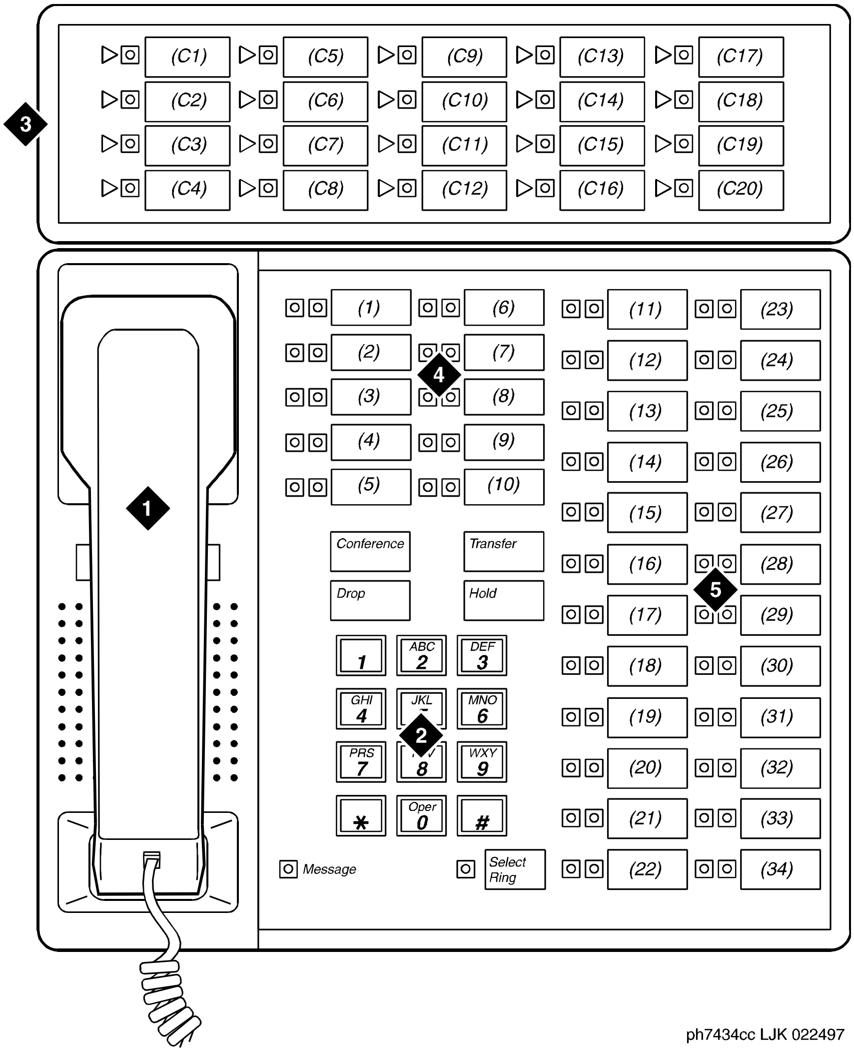


Figure Notes

1. Handset
2. Dial pad
3. Call coverage module with 20 coverage module buttons and status lamps
4. 10 programmable buttons
5. 24 feature buttons (11 to 34)

Figure 42. 7434D telephone with optional call coverage module

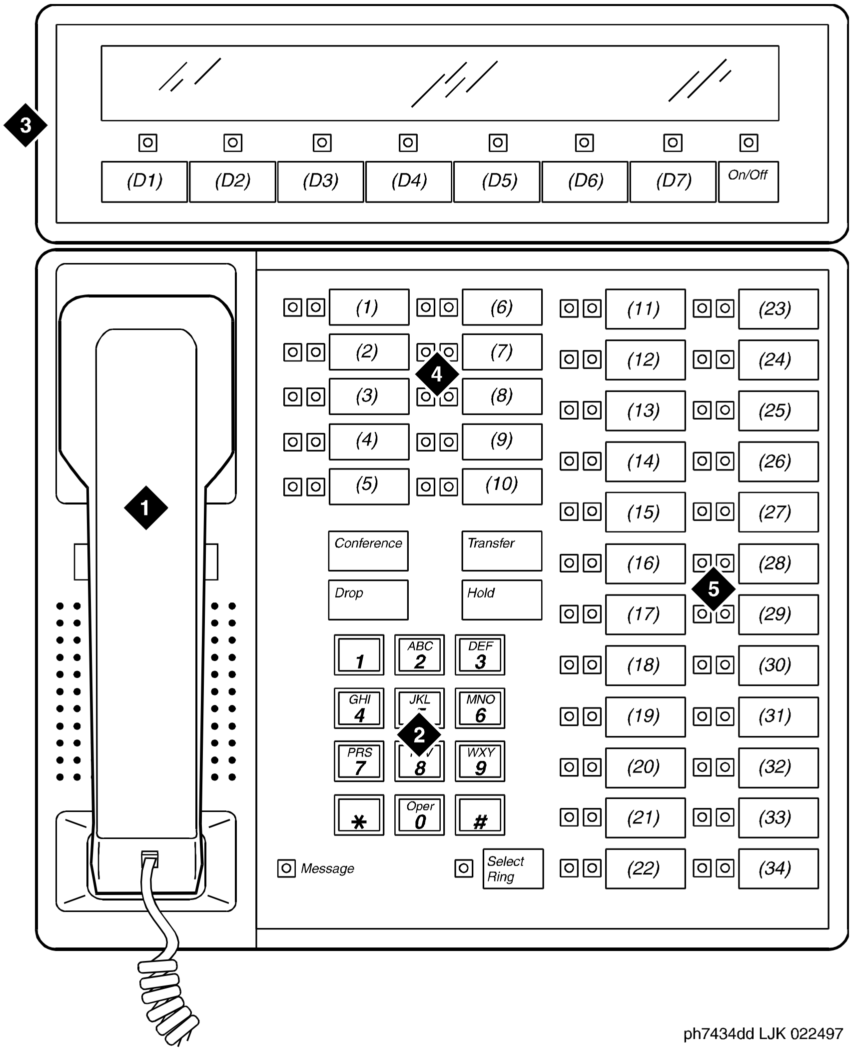


Figure Notes

1. Handset
2. Dial pad
3. Digital display module with 7 display buttons
4. 10 programmable buttons
5. 24 feature buttons (11 to 34)

Figure 43. 7434D telephone with optional digital display module

ISDN telephones (7500s & 8500s)

The Integrated Services Digital Network (ISDN) phones include both the 7500-series and the 8500-series telephones. Each of these phones uses the ISDN communications through a 4-wire "T"-interface.

7505D ISDN-BRI telephone

The 7505D telephone serves as a telephone if it is equipped with a Voice Only Module (VOM). As a telephone, it offers programmable buttons, fixed feature buttons, a Message light, touch-tone dialing, and a built-in, programmable speakerphone or SPOKESMAN loudspeaker.

The 7505 can be equipped with an optional Asynchronous Data Module that provides the user with simultaneous voice and data capabilities. The 7505 equipped with the ADM offers the same voice capabilities as the 7505 equipped with the VOM, plus it allows you to attach data terminals or personal computers to send and receive data through the digital network.

7506D ISDN-BRI telephone

The 7506D serves as a telephone if it is equipped with a Voice Only Module (VOM). As a telephone, it offers programmable buttons, fixed feature buttons, a Message light, touch-tone dialing, and a built-in, programmable speakerphone or SPOKESMAN loudspeaker.

The 7506 can be equipped with an optional Asynchronous Data Module that provides the user with simultaneous voice and data capabilities. The 7506 equipped with the ADM offers the same voice capabilities as the 7506 equipped with the VOM, plus it allows you to attach data terminals or personal computers to send and receive asynchronous data through the digital network.

7507D ISDN-BRI telephone

The 7507D serves as a telephone if it is equipped with a Voice Only Module (VOM). As a telephone it offers programmable buttons, fixed feature buttons, Message light, touch-tone dialing, and a built-in, programmable speakerphone or SPOKESMAN loudspeaker.

The 7507 can be equipped with an optional Asynchronous Data Module that provides the user with simultaneous voice and data capabilities. The 7507 equipped with the ADM offers the same voice capabilities as the 7507 equipped with the VOM, plus it allows you to attach data terminals or personal computers to send and receive data through the digital network.

8503D ISDN-BRI telephones

The 8503T telephone offers: four standard fixed feature buttons, a Message light, three call appearance/flexible feature buttons, each with a red and green light, 12 programmable memory-dialing locations on the dial pad keys, a PROGRAM button for storing numbers at the memory-dialing locations and a MEMORY button for dialing these programmed numbers or codes, the Redial feature, the Mute feature.

8510T ISDN-BRI telephone

The 8510T voice/data telephone offers: 10 call appearance/feature buttons, each with a red and green status light, buttons for the Mute, Redial, Conference, Drop, Transfer, and Hold features (the MUTE, SPEAKER, CONF, and TRANSFER buttons have a red light next to them), a Speakerphone, a Volume control, and an Adjunct jack. You can administer the softkey buttons. Four softkeys and display control buttons below the a 2-line by 24-character display can be used to access such features as a personal Directory, a Call Log, the Self-Test feature, and a personalized ringing pattern for the telephone.

8520T ISDN-BRI telephones

The 8520T voice/data telephone offers the following: 20 call appearance/feature buttons, each with a red and green status light, buttons for the Mute, Redial, Conference, Drop, Transfer, and Hold features (the MUTE, SPEAKER, CONF, and TRANSFER buttons have a red light next to them), a Speakerphone, a Volume control, and an Adjunct jack. Ten softkeys and four display control buttons located on either side of the 7-line by 24-character display can be used to access such features as a personal Directory, a Call Log, the Self-Test feature, and a personalized ringing pattern for the telephone.

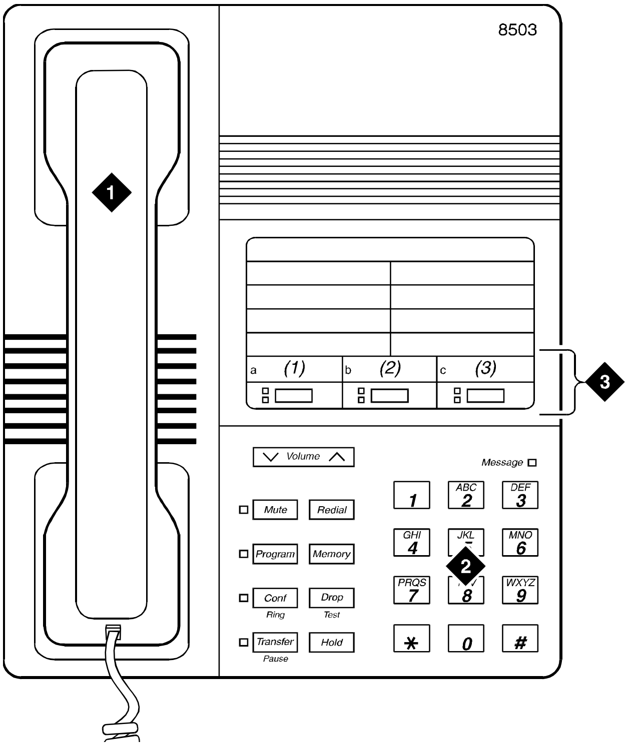


Figure Notes

1. Handset
2. Dial pad
3. 3 programmable buttons

Figure 44. 8503D telephone

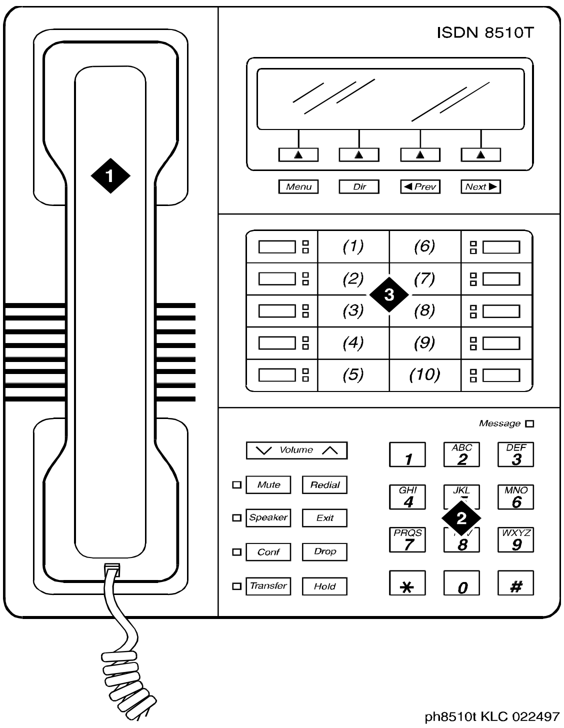


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons

Figure 45. 8510T telephone

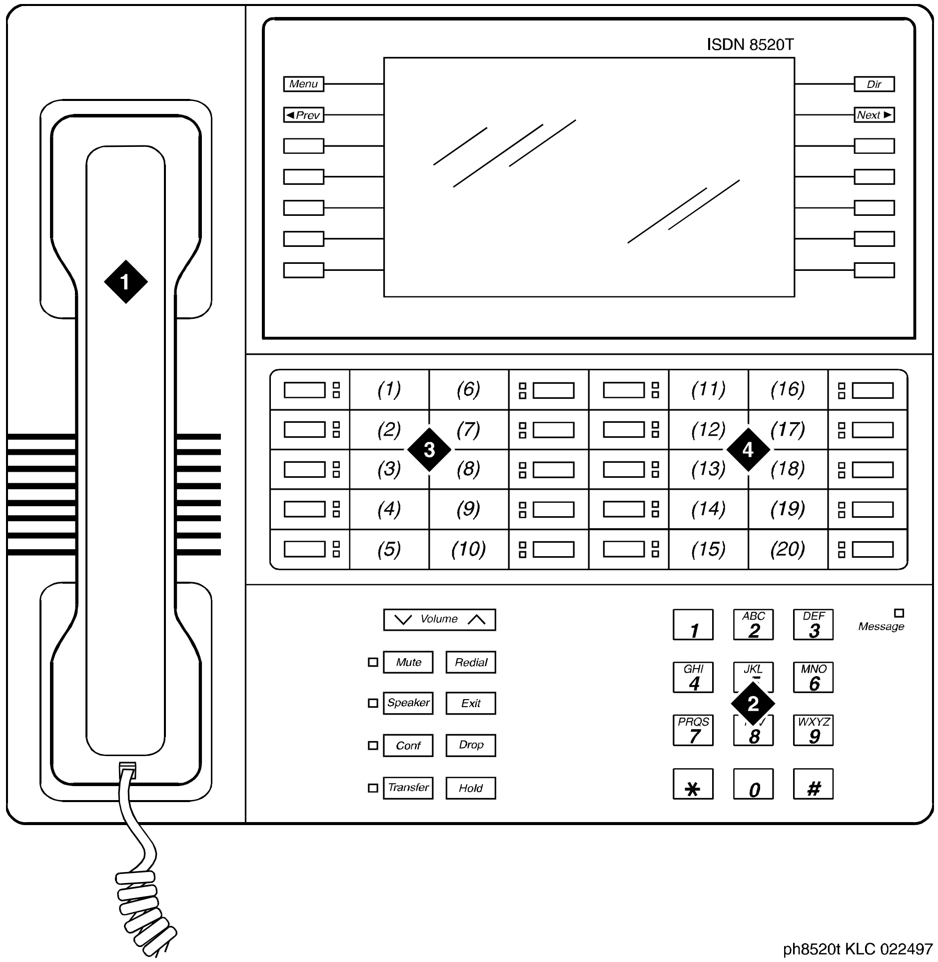


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 10 programmable buttons (11 to 20)

Figure 46. 8520T telephone

NOTE:
The 8520T telephone supports 20 call appearances. The system maximum of 10 call appearance buttons still applies. You can administer the buttons that are not used as call appearance buttons as bridged appearances.

8110 telephones

The basic 8110 (8110A01A, 8110A01B, and 811A01C) and the modified 8110M (8110A01D) telephones are single-line analog telephones. These telephones are exactly the same in appearance: each contains 12 programmable dialing buttons with PROGRAM and PAUSE buttons, automatic redial, a flashing red Message light, and a Hold button. They also have built-in speakerphones with Mute capability and the Automatic Answer feature.

8400-series telephones

8403B telephones

The 8403 telephone is a multi-appearance digital telephone with three call appearance buttons, Conference, Transfer, Drop, and Hold buttons, a TEST button, a blue FEATURE button which allows you to access 12 system features assigned by the System Manager and to choose from among eight different ringing patterns, a MUTE button, a SPEAKER button which accesses a 1-way, listen-only speaker, a red Message light, and a Volume control button.

The 8403 can be used in either a 4-wire or 2-wire environment.

8405B telephone

There are four varieties of the 8405 telephone: the 8405B and 8405B+, the 8405D and 8405D+. All four varieties are multi-appearance digital telephones with five call appearance/feature buttons.The 8405 telephones also have four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a MUTE button, a SPEAKER button, a TEST button, and a Volume control button. The 8405D and 8405D+ allow you to administer 12 softkey feature buttons in addition to the call appearance and feature buttons.

The four 8405 variations have the following differences:

- The 8405B has a 1-way, listen-only speaker, with NO display.
- The 8405B+ has a 2-way speakerphone, without a display
- The 8405D has a 1-way, listen-only speaker and a 2-line by 24-character display.
- The 8405D+ has a 2-way speakerphone and a 2-line by 24-character display.

The 8405 telephones work in 4-wire or 2-wire environments.

8410B telephone

The 8410 telephone is a multi-appearance digital telephone with 10 call appearance/feature buttons, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a MUTE button, a SPEAKER button which can access either a 2-way speakerphone or a 1-way, listen-only speaker, a TEST button, and a Volume control button.

- The 8410B is the basic set, without a display.
- The 8410D (8410D03A) has a built-in 2-line by 24-character display. Those users who have an 8410D with display can access 12 features with the softkeys and display control buttons. These 12 features can be used *in addition to* the features on the call appearance/feature buttons.

The 8410 telephone can work in both 4-wire and 2-wire environments.

8411B and 8411D telephones

The 8411 telephone is a multi-appearance digital telephone with 10 call appearance/feature buttons, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a blue SHIFT button, a MUTE button, a SPEAKER button which can access either a 2-way speakerphone or a 1-way, listen-only speaker, a TEST button, and a Volume control button.

The rear of the 8411 telephone has two jacks: The Analog Adjunct jack can be used for connecting answering machines, fax machines, PC or laptop data/fax modem cards, data sets or modems, audio teleconferencing equipment, and TTY machines commonly used by the hearing impaired. The RS-232-D Jack can be used for connecting the telephone to a COM port on an IBM®-compatible personal computer on which you can load PassageWay Solution software.

There are two varieties of the 8411 telephone: the 8411B (8411D02A) is the basic set, without a display; the 8411D (8411D01A) has a built-in 2-line by 24-character display. Those users who have an 8411D with display can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features on the call appearance/feature buttons.

The 8411 telephone can work in both 4-wire and 2-wire environments.

8434D telephone

The basic 8434 (8434D01A) and the enhanced 8434DX (8434D02A) telephones are multi-appearance digital telephones which offer 34 call appearance/feature buttons, each with a red light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a MUTE button, a SPEAKER button which accesses either a 2-way speakerphone or a 1-way listen-only speaker, a TEST button, a SHIFT button (some 8434DX telephones will have a RING button instead), a red Message light, personalized ringing, a built-in speakerphone with a reset option, and a built-in 2-line by 40-character VFD display. The 8434 and 8434DX also have five softkeys and four display control buttons which allow the user to access 15 features. These softkey features can be used *in addition to* the features on the call appearance/feature buttons.

The 8434 and 8434DX telephones can be used in both a 4-wire and a 2-wire environment.

NOTE:
In order to use the display on the 8434 or 8434DX telephone and to use an 801A expansion module connected to the 8434DX, you must connect an auxiliary power supply to the telephone.

You can connect an 801A Expansion Module to the 8434DX telephone to provide 24 additional call appearance/feature buttons.

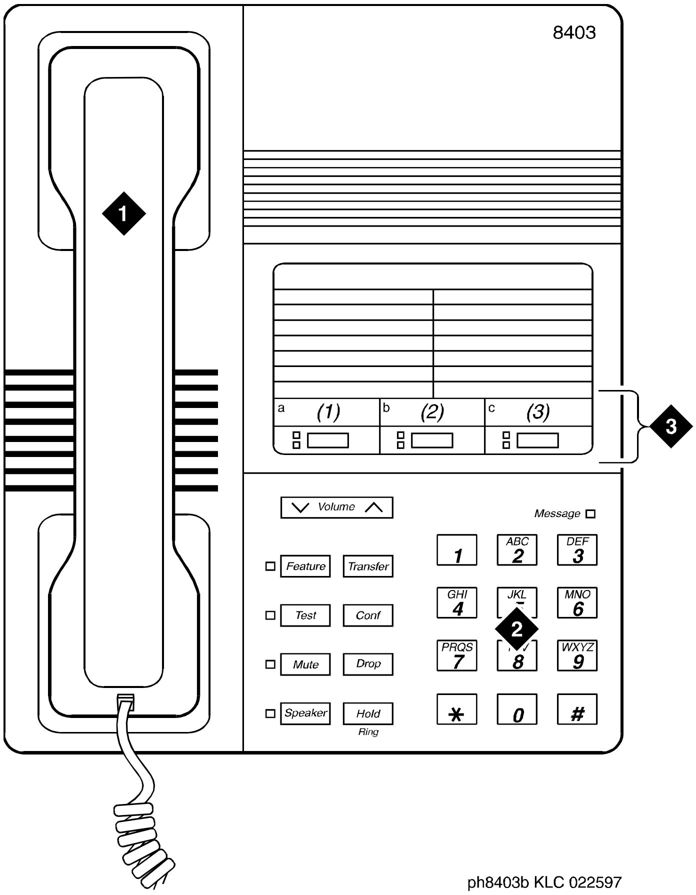


Figure Notes

1. Handset
2. Dial pad
3. 3 programmable buttons

Figure 47. 8403B telephone

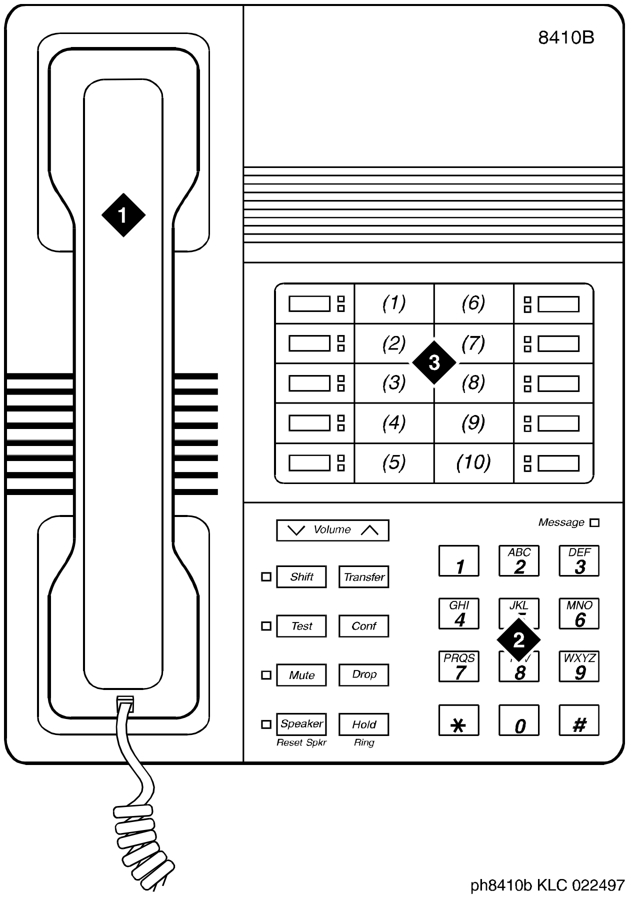


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons

Figure 48. 8410B telephone

NOTE:
The 8405B and 8405B+ look like the 8410B with the exception that the 8405 series do not have the second column of line appearances.

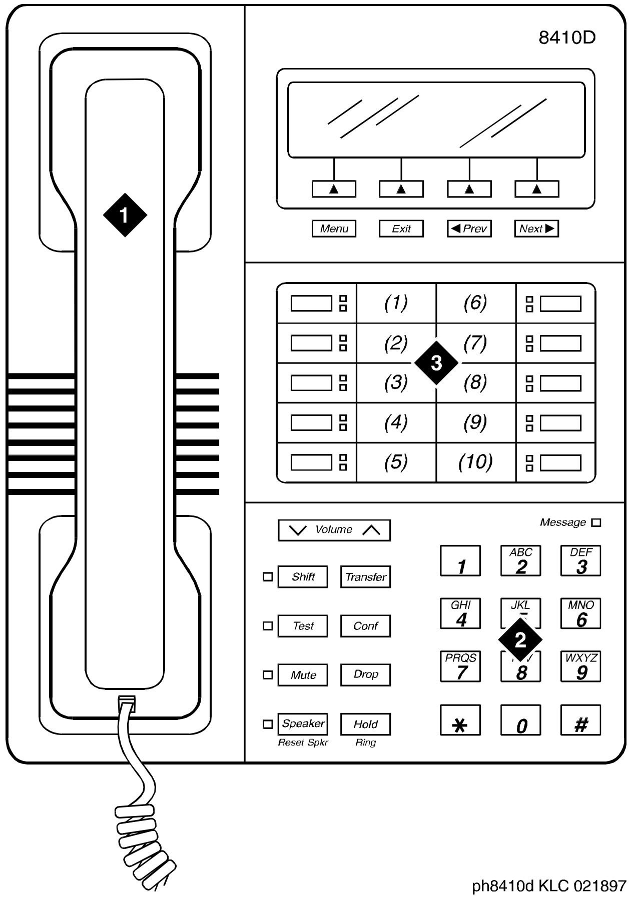


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons

Figure 49. 8410D telephone

NOTE:
The 8405D and 8405D+ look like the 8410D with the exception that the 8405 series do not have the second column of line appearances.

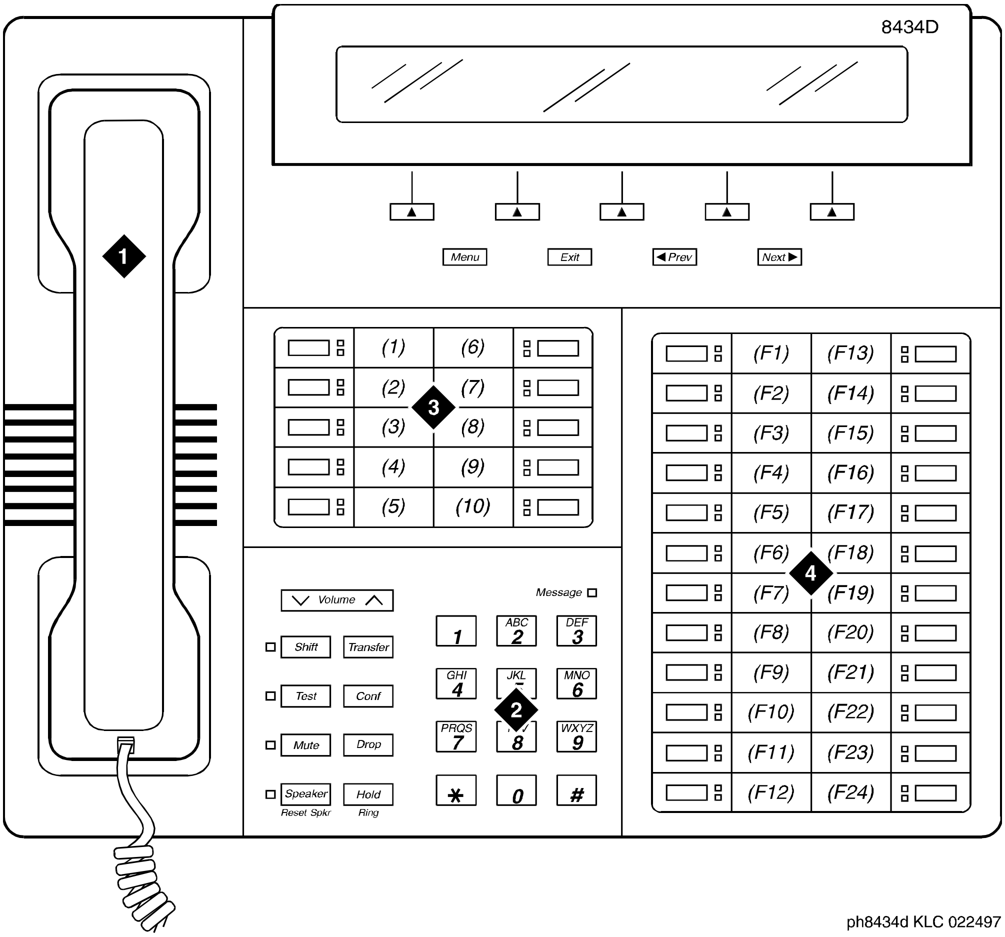


Figure Notes

1. Handset
2. Dial pad
3. 10 programmable buttons
4. 24 feature buttons

Figure 50. 8434D telephone

CALLMASTER telephones

There are several types of CALLMASTER telephones:

- 602A and 602D CALLMASTER

  The 602 CALLMASTER models have a display, a Message light, a Mute button, and four fixed feature buttons: Conference, Drop, Hold, and Transfer. You can administer its 10 call appearance/feature (2-lamp) buttons and its 17 feature-only (1-lamp) buttons.

- 603D (CALLMASTER II)

  The CALLMASTER II model has a display, a Message light, and the Mute, Select, Log In, and Release buttons. It also has four fixed features: Conference, Drop, Hold, and Transfer. You can administer its 6 call appearance/feature (2-lamp) buttons and its 15 feature-only (1-lamp) buttons.

- 603E (CALLMASTER III)

  The CALLMASTER III model has a display, a Message light, and the Select, Mute, Log In, and Release buttons. It also has four fixed features: Conference, Drop, Hold, and Transfer. You can administer its 6 call appearance/feature (2-lamp) buttons and its 15 feature-only (1-lamp) buttons. Note that you can assign any feature to the Log In and Release buttons.

  You can connect the CALLMASTER III to either a standard 4-wire DCP or a 2-wire circuit pack.

- 603F (CALLMASTER IV)

  The CALLMASTER IV model has a display, a Message light, and the Select, Mute, Log In, and Release buttons. It also has four fixed features: Conference, Drop, Hold, and Transfer. You can administer its 6 call appearance/feature (2-lamp) buttons and its 15 feature-only (1-lamp) buttons. Note that you can assign any feature to the Log In and Release buttons.

  You can connect the CALLMASTER IV to either a standard 4-wire DCP or a 2-wire circuit pack.

- 606A (CALLMASTER VI)

  The CALLMASTER VI model is a miniature, 8-button, 2-headset jack, digital telephone that is controlled by the user’s personal computer (PC) through an RS-232 serial-port connection.

- 607A (CALLMASTER V)

  The CALLMASTER V model has a display, softkeys, and the display control buttons (Menu, Exit, Previous, and Next). This model does not have a standard handset, but you can connect a handset to one of its headset jacks. The CALLMASTER V has six fixed feature buttons: Speaker, Mute, Hold, Redial, Conference, and Transfer. You can administer its 16 call appearance/feature (2-lamp) buttons, however, one of these buttons must be administered as a Headset On/Off button and a second one must be administered as a Release button. You can also administer the 12 softkey buttons.

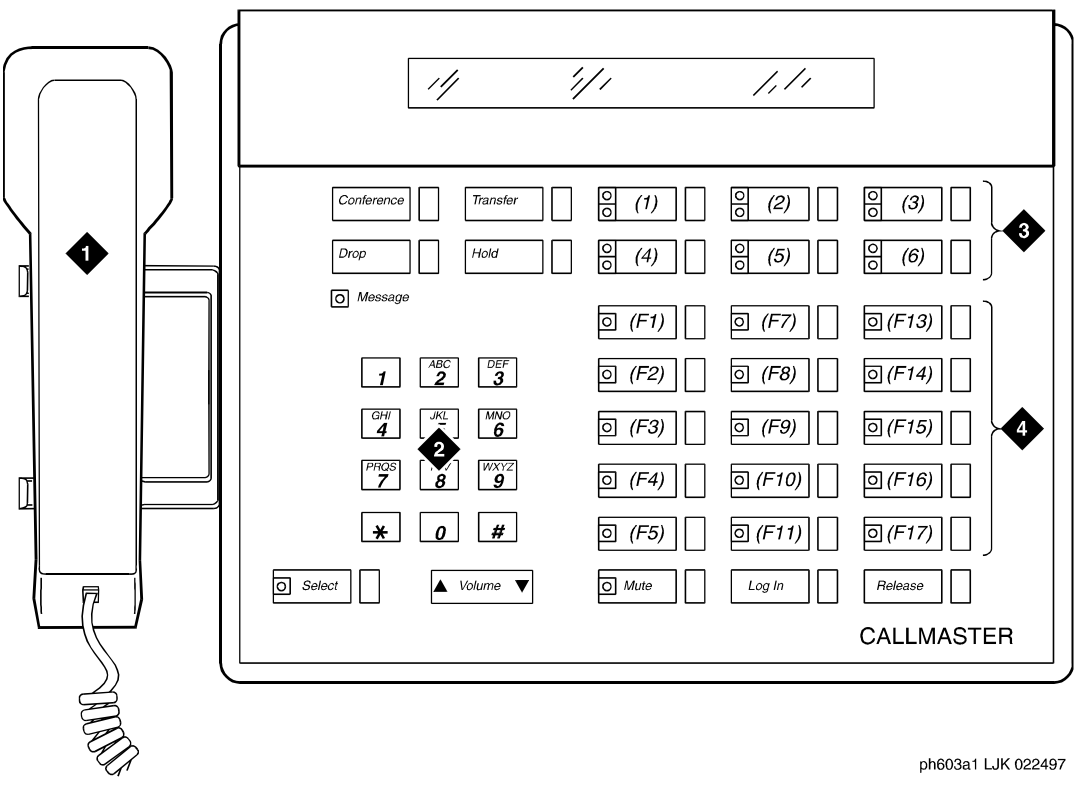


Figure Notes

1. Handset
2. Dial pad
3. 6 programmable buttons
4. 18 feature buttons

Figure 51. CALLMASTER II/CALLMASTER III digital telephone

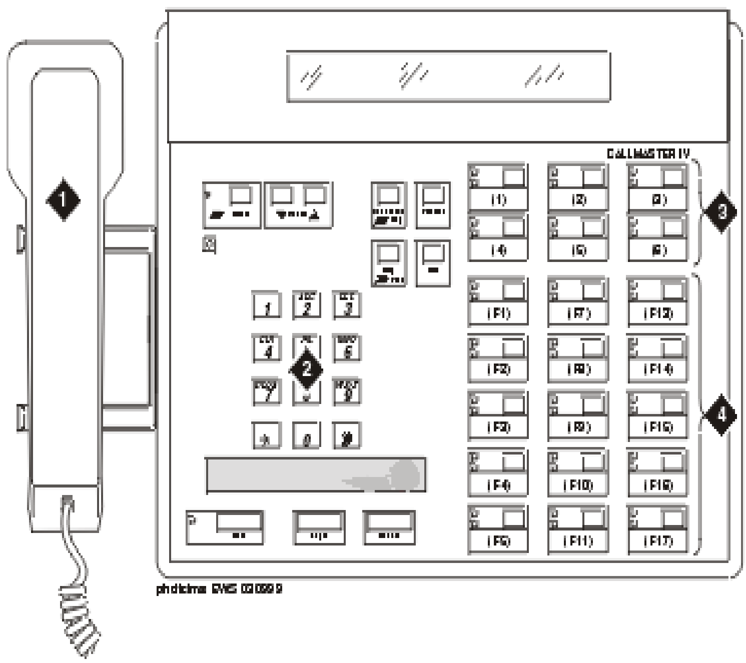


Figure Notes

1. Handset
2. Dial pad
3. 6 programmable buttons
4. 15 feature buttons

Figure 52. CALLMASTER IV digital telephone

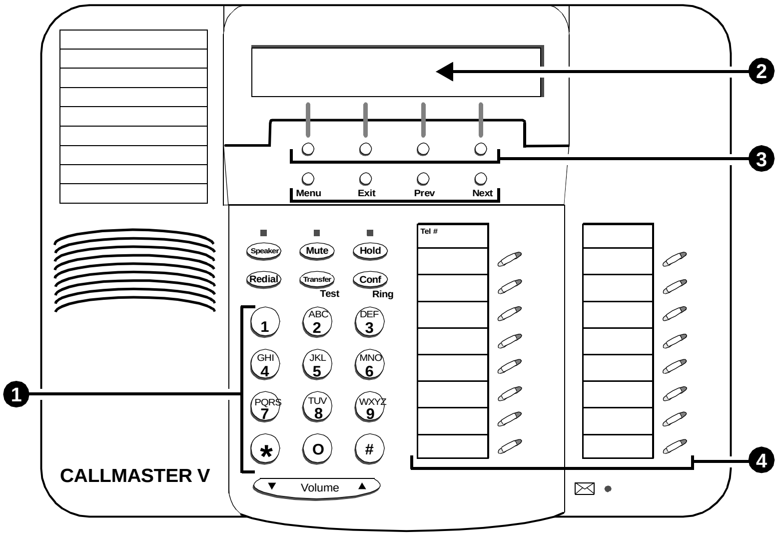


Figure Notes

1. Dial pad
2. Display
3. 4 softkey buttons
4. 16 call appearance/feature buttons

Figure 53. CALLMASTER V digital telephone

Cordless telephone

MDC9000 cordless telephone

The MDC 9000 Cordless Telephone has two basic parts, the handset and the charging base.

- The handset has line/programmable feature/intercom buttons, Conference, Drop, Hold, Transfer, Status, and Feature buttons, Headset On/Off and Handset On/Off buttons, a LCD display, an earpiece volume control switch, battery charging contacts, a directory card, and a headset jack.
- The charging base has a handset hook, ringer volume controls, battery charging contacts, a Talk indicator, a Charge indicator, a Message indicator, and a base ringer.

MDW9000 cordless telephone

The MDW 9000 Wireless Telephone is part of the TransTalk™ 9000 Digital Wireless System family of telephones. This wireless telephone has three basic parts, the handset the charging cradle, and the radio module.

- The handset has line/programmable feature/intercom buttons, Drop, Mute, Hold, Conference, and Transfer buttons, a Headset On/Off button and a Handset On/Off button, a LCD display, a Volume control switch, battery charging contacts, a flexible antenna, and a Headset jack.
- The charging cradle has a handset hook, a spare battery cover, a spare battery.
- The radio module has Power, Pass, and Radio indicator lights, a top hook, card edge, and snap lock which connect the radio module to the carrier assembly/backplane, an antenna, and power plug and line connectors.

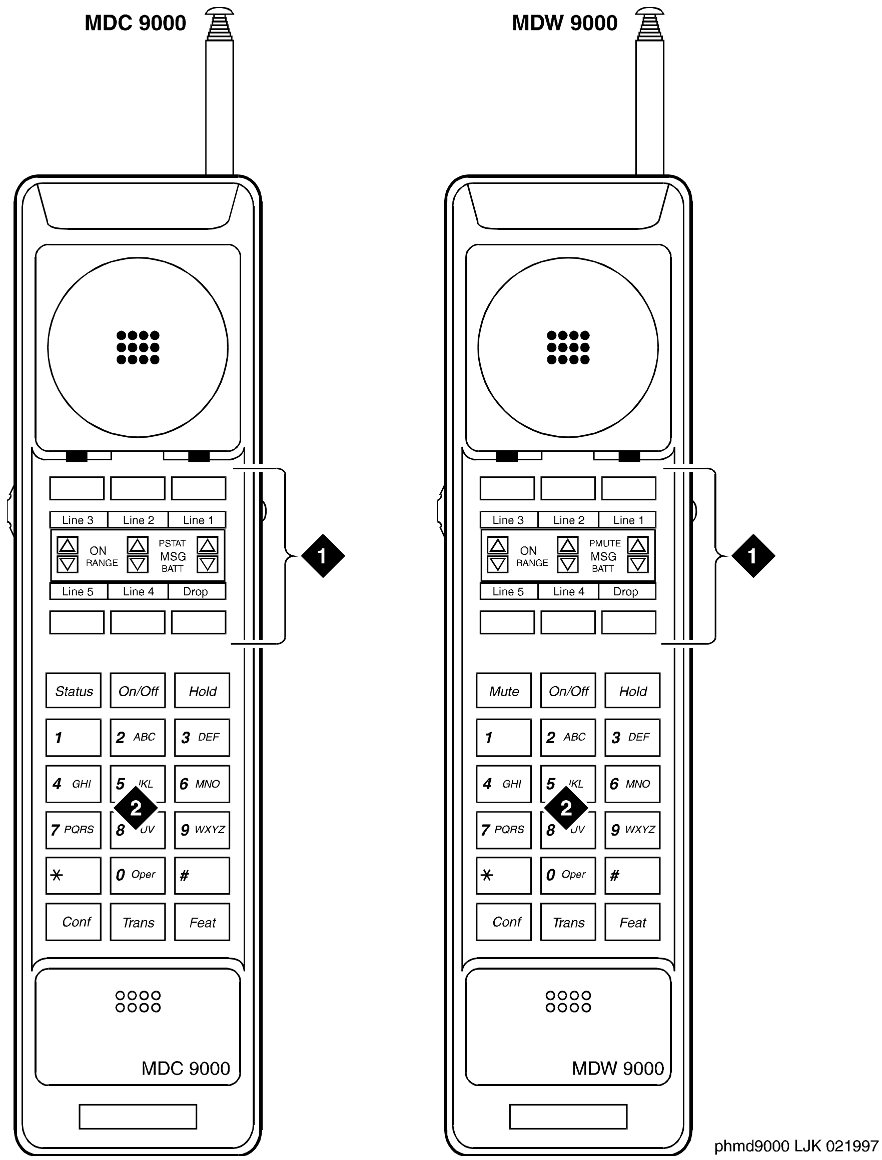


Figure Notes

1. 6 programmable buttons
2. Dial pad

1. 6 programmable buttons
2. Dial pad

Figure 54. MDC9000 and MDW9000 cordless telephones

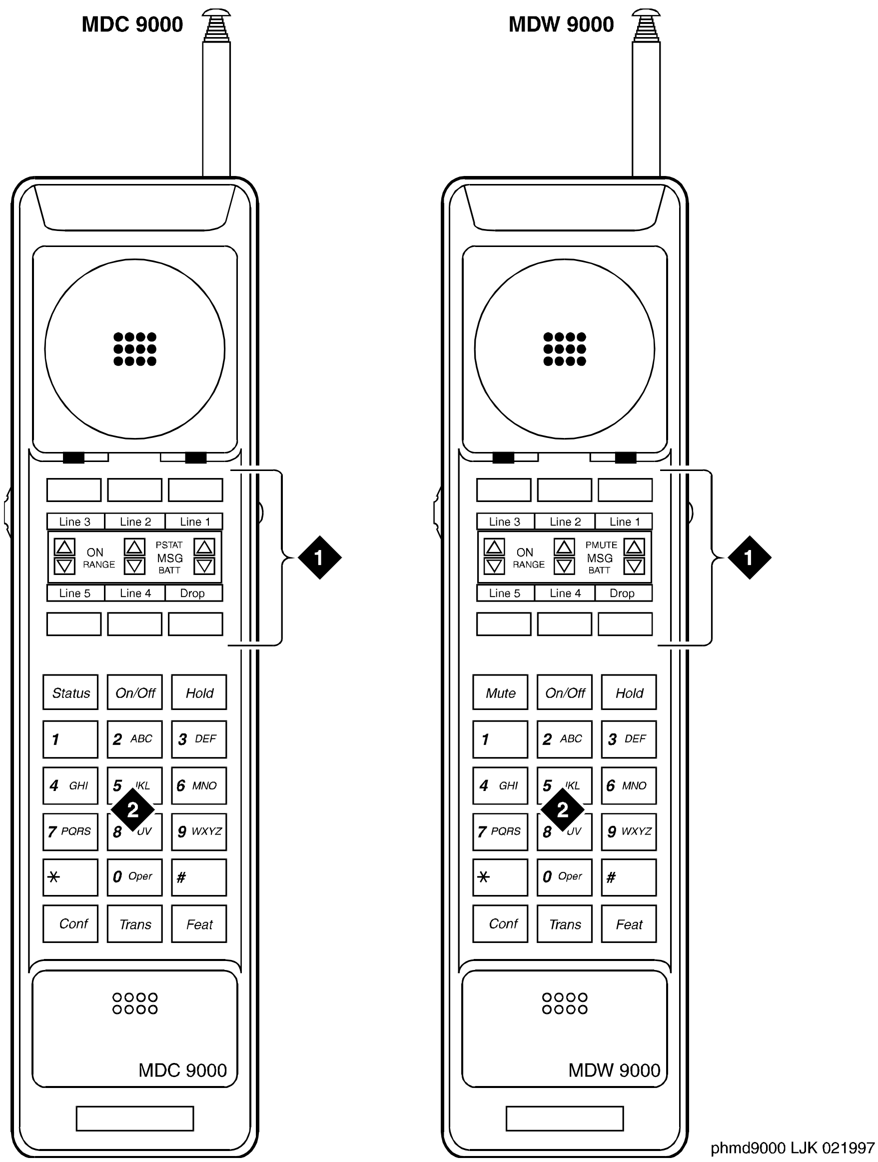


Figure Notes

MDC 9000	MDW 9000
1. 6 programmable buttons	1. 6 programmable buttons
2. Dial pad	2. Dial pad

Figure 55. MDC9000 and MDW9000 cordless telephones

Internet Protocol (IP) Softphones

IP Softphones extend the level of MultiVantage services. They turn a PC or a laptop into an advanced telephone. IP Softphones allow you to send voice and fax from the MultiVantage through an Internet Protocol (IP) network to other MultiVantage systems that have this feature. You can place calls, take calls, and handle multiple calls on your PC.

IP Softphones extend MultiVantage multifunction, multiline features support to IP-connected endpoints (typically user PCs). With certain exceptions, every feature available for wired-endpoint voice calling is available for IP-based calling; it supports full internetworking with conventional circuit-switched stations and trunks.

There are three IP Softphone applications available: the road-warrior, telecommuter, and the stand-alone H.323. Avaya IP Agent is a modified telecommuter configuration that uses the Avaya IP Agent interface, rather than the Avaya IP Softphone interface.

Road-warrior application

The road-warrior application enables travelers to use MultiVantage features from temporary remote locations, such as a hotel room. The road-warrior configuration uses two separate software applications running on a user's PC that is connected to a MultiVantage system over an IP network. The single network connection carries two channels: one for call control signaling and one for voice. IP Softphone software handles the call signaling and an H.323 V2-compliant audio application (such as Microsoft® NetMeeting®) handles the voice communications.

Telecommuter application

The telecommuter application enables remote workers to use MultiVantage features from a remote location, such while telecommuting from a home office. The telecommuter configuration uses two connections to the MultiVantage system: a connection to the PC over the IP network and a connection to the telephone over the public-switched telephone network (PSTN). The PC user places and takes calls with the IP Softphone interface and uses the telephone handset to speak and listen.

You can also use a variation of the telecommuter application for call center agents: Avaya IP Agent. This application uses the Avaya IP Agent interface instead of the IP Softphone interface to emulate a remote CallMaster phone.

Stand-alone H.323

The stand-alone H.323 application enables travelers to use some MultiVantage features from a remote location. This application uses a PC running an H.323 v2-compliant audio application, such as Microsoft NetMeeting. The H.323 application controls the call signaling and the voice path. However, since it does not use the IP Softphone interface, this configuration is capable of operating only as a single-line telephone without any additional assigned features. You can provide stand-alone H.323 users only features that can they can activate with dial access codes.

Related topics

For instructions on how to administer an IP Softphone on your system, refer to ''Adding an IP Softphone'' on page 109.

You can also find information on the IP Softphone CD (refer to *IP Softphone Overview and Troubleshooting* and *Getting Started*).

Features and technical reference

21

AAR and ARS partitioning

You can use Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) partitioning to change the call routing plan for up to 8 different user groups within a single MultiVantage system. You assign a Partition Group Number (PGN) to each user group and identify different call routing treatment for each PGN.

For example, you can partition hotel employees and guests into separate groups (PGN) and route the calls differently. When a guest makes a long-distance call, the guest's PGN and digit analysis tables route the call to a telephone-billing system that allocates long-distance charges. A similar call placed by an employee routes over a direct-distance dialing (DDD) trunk.

Detailed description

Partition user groups are used only with AAR, and ARS, and Uniform Dial Plan (UDP). You can assign AAR and ARS partitioning to phones, attendant consoles, remote-access users, data endpoints, and incoming trunks.

Use partitioning for:

- groups with different routing due to special billing needs
- groups that have dedicated use of a particular network facility
- groups in different businesses serviced by a single system
- data users who require special facility types on outgoing calls

You can assign a route pattern to just one partitioned user group or you can assign a route pattern to all your partitioned user groups.

You assign the PGN on the Class of Restriction (COR) screen, and then assign the COR on each station screen. When a user dials an AAR or ARS feature access code and a number, the switch uses the PGN of the caller's COR to determine the route pattern. The PGN field appears on the COR screen only if Time of Day Routing is **n** on the System Parameters Customer Options screen.

If Time of Day Routing is **y** on System Parameters Customer Options, you specify the partition group number (PGN) on the Time of Day Routing Plan screen. Refer to ''Time of Day Routing'' on page 1766 for more information.

Interactions

- Bridged Call Appearance

  If a Bridged Call Appearance is used for an AAR or ARS call, the system uses the bridged extension's PGN instead of the caller's PGN.

- DCS

  When a call routes over DCS, the far-end switch uses the incoming trunk's PGN to route the call.

- Remote Access

  When a remote-access user dials barrier code or authorization code and an ARS feature access code, the barrier code or authorization code's COR determines the PGN.

- Straightforward Outward Completion and Through Dialing

  If the attendant assists or extends a call and dials an ARS feature access code, the attendant's COR determines the PGN if the individual extension is assigned. Otherwise, the COR set on the console parameter determines the PGN.

Related topics

Refer to''Defining ARS Partitions'' on page 261 to see how to set up an ARS partition group.

Refer to ''Setting up time of day routing'' on page 264 to see how to set up Time of day routing.

Abbreviated Dialing

Abbreviated Dialing (AD) provides easy access to selected numbers by reducing the number of digits users have to dial to place a call. Instead of dialing the entire number, the user dials a short code to access the number. The system then dials the stored number automatically. You can assign abbreviated dialing buttons to phones, so users can dial frequently-called numbers by pressing a single button.

You can also assign privileged numbers to abbreviated dialing lists, so you can allow a user to place calls to numbers that might otherwise be restricted.

SECURITY ALERT:
Privileged group-number, system-number, and enhanced-number lists provide access to numbers that typically would be restricted.

List types

You can store phone numbers in 4 different types of abbreviated dialing lists:

- personal
- group
- system
- enhanced

Your switch type and version determines which lists you have available and how many entries you can have on each list. You can assign up to 3 AD lists to each user (extension). The 3 lists can be made up of any combination of a system list, an enhanced list, up to 3 personal lists, or up to 3 group lists. Each abbreviated dialing entry can have up to 24 characters.

Personal lists

You can provide personal lists to phone users who need their own set of stored numbers. You determine which users have access to a personal list and the size of each list. Either you or the user can assign phone numbers to personal lists. A personal list is created automatically when you assign the list to an individual phone. Each user can have up to 3 personal lists. Note that you cannot assign personal list to the attendant.

Group lists

You can define group lists for groups or departments, such as purchasing or human resources, where members of the group have the need to frequently dial the same numbers. You determine which users have access to group lists and each user may have access to up to 3 group lists. You can program the list or you can designate a user in each group to program the list. You specify this designated user on the Abbreviated Dialing Group List screen.

System lists

You can define one system list for the entire organization. Most administrators assign this list to every phone and allows everyone in the organization to use the list. If you choose to let everyone use the system list, you should only add numbers to the list that anyone in your organization may call. For example, you may want to add an emergency phone number or phone numbers for other office locations to this list.

The system list can contain up to 100 entries and can only be changed by a system administrator.

Enhanced lists

Enhanced-number lists are used by telephone users, data-terminal users, and attendants who need more list entries than those allowed in group-number and system-number lists. One enhanced-number list is allowed per system in addition to the system-number list. The enhanced-number list can contain any number or dial-access code. You administer the enhanced-number lists and determine which users can access the list.

Considerations

- You cannot remove a telephone or attendant if it is designated as the extension number that is permitted to program a group-number list.
- When using an AD button to access a messaging system, the user’s login and password should not be assigned to the button. The system ignores button entries after the messaging access number.
- You can use an abbreviated dialing list at any time during incoming or outgoing calls.

Interactions

- Last Number Dialed

  The Last Number Dialed feature redials the same number a user just dialed, even if the user accessed an abbreviated dialing list for the previous call. However, if the last dialed string includes any special characters (such as indefinite wait, mark, pause, suppress, or wait) these characters are ignored by last-number-dialed call.

  If the previously-called number was in an AD privileged list, and if the user is not normally allowed to dial the number because of his or her class of restriction, they cannot redial the number using Last Number Dialed. To redial the number, the user must again access the AD privileged list.

Related topics

Refer to ''Adding abbreviated dialing lists'' on page 151 for instructions on creating abbreviated dialing lists.

Refer to ''Abbreviated Dialing List'' on page 567 for detailed descriptions of the fields on the abbreviated dialing screens.

Access security gateway

Access Security Gateway (ASG) prevents unauthorized access by requiring the use of the ASG Key for logging into the system. The ASG Key can be:

- a hand-held device, or
- a software module you load on the PC you use for accessing the system.

Detailed Description

Authentication is successful only when MultiVantage and ASG communicate with a compatible key. The challenge/response negotiation starts after establishing an RS-232 session and you enter a valid MultiVantage login ID. The authentication transaction consists of a challenge, issued by MultiVantage and based on the login ID entered by you, followed by the expected response, again entered by you. The core of this transaction is a secret key, which is information-possessed by both the lock (ASG) and the key. Interception of either the challenge or response during transmission does not compromise the security of the system. The relevance of the authentication token used to perform the challenge/response is limited to the current challenge/response exchange (session).

NOTE:
ASG does not protect login access to a Multiple Application Platform for DEFINITY (MAPD).

The supported key consists of a hand-held encryption-generating device (ASG Key). The key (response generator) device is pre-programmed with the appropriate secret key to communicate with corresponding ASG protected login IDs on MultiVantage.

The *Avaya Products Security Handbook* contains information about:

- toll fraud and what you can do to prevent it.
- methods people use to gain access to your system, how to detect toll fraud, and what to do if you suspect that your system has been compromised.
- security information for many Avaya products, so you can be sure that all of your telecommunications equipment is secure.
- security checklists for each of these products. You should go through these with your Avaya representative for each piece of equipment you use.

Interactions

- Customer Access to INADS Port

  If access to the INADS port is disabled on a system-wide basis, administering access to the SYSAM-RMT or INADS port, through ASG, does not override the INADS port restriction. Administration does not prohibit assignment of ASG to the SYSAM-RMT or INADS port. However, in a configuration where this method of access is blocked, you will be denied access to the system through the SYSAM-RMT or INADS port even if you attempt to access the port using a valid ASG login ID.

  If access to the INADS port has been disabled on a login basis, administering access to the SYSAM-RMT or INADS port, via ASG, will not override the INADS port restriction.

- Login Administration

  The standard user interface for MultiVantage login administration has not been modified by ASG. Also, the standard MultiVantage login user interface is maintained in cases where ASG parameters have not been administered for the login and/or port.

- Security Violation Notification

  ASG does not support an interface to SVN. Session rejection events do not appear in the monitor security-violations login report and referral calls are not spawned in the event that the number of rejected ASG sessions exceeds the threshold/time interval criteria imposed by SVN.

- Security Measurements

  ASG session establishment or reject events do not increment the Successful Logins, Invalid Attempts, Invalid IDs, Forced Disconnects, Login Security Violations or Trivial Attempts counters maintained for the list measurements security-violations detail report. Additionally, login specific information maintained by the measurements security-violations summary report does not include ASG related data.

Related topics

Refer to ‘‘Using access security gateway’’ on page 387 for instructions.

Administered Connections

An Administered Connection (AC) is a connection between 2 access or data endpoints. MultiVantage automatically establishes the connection based on the attributes you administer. An AC provides the following capabilities:

- Support of both permanent and scheduled connections
- Auto Restoration (preserving the active session) for connections routed over Software Defined Data Network (SDDN) trunks
- Administrable retry interval (from 1 to 60 minutes) per AC
- Administrable alarm strategy per AC
- Establishment/retry/auto restoration order based on administered priority

Detailed description

The SDDN provides virtual private-line connectivity using the AT&T Switched Network. Access to the service is provided via an Integrated Services Digital Network (ISDN) trunk group whose Service Type field is **SDDN**. The system uses the Destination field on the Administered Connection screen to route calls when AC is active, based on associated authorized time-of-day fields.

Establish an AC between the following:

- Two endpoints on the same switch
- Two endpoints in the same private network, but on different switches
- One endpoint on the controlling switch and another endpoint off the private network

In all configurations, administer the AC on the switch that has the originating endpoint. For an AC in a private network, if the two endpoints are on two different switches, normally the connection routes via Automatic Alternate Routing (AAR) through tie trunks (ISDN, DS1, or analog tie trunks) and intermediate switches. If required, route the connection via Automatic Route Selection (ARS) and Generalized Route Selection (GRS) through the public network. The call routes over associated ISDN trunks. When the far-end answers, a connection occurs between the far-end and the near-end extension in the Originator field on the Administered Connection screen.

Access endpoints

Access endpoints are non-signaling trunk ports. They neither generate signaling to the far-end of the trunk nor respond to signaling from the far-end. Designate an access endpoint as the originating endpoint or destination endpoint in an AC.

Typical AC applications

The following are typical AC applications:

- A local data endpoint connection to a local or remote-access endpoint. Examples: a modular processor data module (MPDM) ACCUNET digital service connecting to SDDN via an ISDN trunk-group DS1 port; an MPDM ACCUNET digital service connecting to an ACCUNET Switched 56 Service via a DS1 port.
- A local-access endpoint connecting to a local or remote-access endpoint. Examples: a DSO cross-connect and a 4-wire leased-line modem to a 4-wire modem connection via an analog tie trunk.
- A local data endpoint connecting to a local or remote data endpoint such as a connection between two 3270 data modules.

NOTE:
The following guidelines do not include AAR and ARS, or GRS administration information for routing AC calls over trunk groups. See the respective feature elsewhere in this book for that information.

Establishing Administered Connections

The originating switch attempts to establish an AC only if one of the following conditions exist:

- AC is active.
- AC is due to be active (either a permanent AC or time-of-day requirements are satisfied if it is a scheduled AC).
- Originating endpoint is in-service or idle state.

If the originating endpoint is not in service or is idle, no activity takes place for the AC until the endpoint transitions to the desired state. The originating switch uses the destination address to route the call to the desired endpoint. When the switch establishes 2 or more ACs at the same time, the switch arranges the connections in order of priority.

AC attempts can fail for the following reasons:

- Resources are unavailable to route to the destination.
- A required conversion resource is not available.
- Access is denied by Class of Restriction (COR), Facilities Restriction Level (FRL), or Bearer Capability Class (BCC). Or, an attempt is made to route voice-band-data over SDDN trunks in the public switch network.
- Destination address is incorrect.
- Destination endpoint is busy.
- Other network or signaling failures occur.

In the event of a failure, an error is entered into the error log, which generates an alarm, if it is warranted by your alarming strategy. You can display AC failures via the **status administered-connection** command.

As long as an AC is due to be active, the originating switch continues to establish an AC unless the attempt fails because of an administrative error (for example, a wrong number) or service-blocking condition (for example, outgoing calls barred).

- The frequency with which failed attempts are retried is determined by the administered retry interval (1 to 60 minutes) for each AC.
- Retries are made after the retry interval has elapsed regardless of the restorable attribute of the AC.
- ACs are retried in priority order.
- When you change the time of day on the switch, an attempt is made to establish all ACs in the waiting-for-retry state.

Dropping Administered Connections

An AC remains active until one of the following occurs:

- The AC is changed, disabled, or removed.
- The time-of-day requirements of a scheduled AC are no longer satisfied.
- One of the endpoints drops the connection. This could be because of user action (in the case of a data endpoint), maintenance activity resulting from an endpoint failure, busying out of the endpoint, or handshake failure. If the endpoints are incompatible, the connection is successful until handshake failure occurs.

  NOTE:
  An AC between access endpoints remains connected even if the attached access equipment fails to handshake.

- An interruption (for example, facility failure) occurs between the endpoints.

If an AC drops because it was disabled/removed or is no longer due to be active, no action is taken. If an AC drops because of changed AC attributes, an immediate attempt is made to establish the connection with the changed attributes if it is still due to be active. Existing entries in the error/alarm log are resolved if they no longer apply. If handshake failure resulted in the dropping of the connection, in the case of an AC involving at least one data endpoint, no action is taken for that AC until the change administered-connection command is executed.

Administered Connections failure: Auto Restoration and Fast Retry

When an active AC drops prematurely, you must invoke either auto restoration or fast retry to determine whether auto restoration is attempted for an active AC.

If you option AC for auto restoration and the connection was routed over SDDN trunks, auto restoration is attempted. During restoration, connections are maintained between the switch and both endpoints. In addition to allowing the active session to be maintained, AC also provides a high level of security by prohibiting other connections from intervening in active sessions. Auto restoration generally completes before the 60-second endpoint holdover interval. If auto restoration is successful, the call might be maintained (no guarantee). The restoration is transparent to the user with the exception of a temporary disruption of service while restoration is in progress. A successful restoration is reflected by the *restored* state on the status Administered Connection screen. Although the restoration was successful, the data session may not have been preserved.

If auto restoration is not active or if the AC is not routed over SDDN trunks, the switch immediately attempts to reestablish the connection (fast retry). The switch also attempts a retry if the originating endpoint initiated the drop. With fast retry, connections are not maintained on both ends. Fast retry is not attempted for an AC that was last established via fast retry, unless the AC is active for at least two minutes.

If auto restoration or fast retry fails to restore or reestablish the connection, the call drops and the AC goes into retry mode. Retry attempts continue, at the administered retry interval, as long as the AC is due to be active.

Interactions

- Abbreviated Dialing

  Use Abbreviated Dialing entries in the Destination field on the Administered Connection screen. Entries must comply with restrictions.

- Busy Verification of Stations and Trunks

  This feature does not apply to access endpoints because they are used only for data.

- Call Detail Recording

  For an AC that uses a trunk when CDR is active, the origination extension is the originator of the call.

  CDR is not available for access endpoints.

- Class of Restriction

  Reserve a COR for AC endpoints and SDDN trunks. This restricts endpoints that are not involved in AC from connecting to SDDN trunks or endpoints involved in AC.

- Class of Service/Call Forwarding

  Assign to an AC endpoint a COS that blocks Call Forwarding activation at the endpoint.

- Data Call Setup

  Do not assign a default dialing destination to a data module when it is used in an AC.

- Data Hotline

  Do not assign a hotline destination to a data module that is used in an AC.

- Digital Multiplexed Interface (DMI)

  Use DMI endpoints as the destination in an AC. DMI endpoints do not have associated extensions, so do not use them as the originator in an AC.

- Facility Test Calls

  The feature does not apply to access endpoints because an access endpoint acts as an endpoint rather than as a trunk.

- Hunting

  Do not use a hunt-group extension as the origination extension of an AC.

- Modem Pooling

  If you require a modem in an AC, one is inserted automatically. If no modem is available, the connection drops.

- Non-Facility Associated Signaling (NFAS) and D-Channel Backup

  Auto restoration for an AC that initially is routed over an NFAS facility may fail if the only backup route is over the facility on which the backup D-channel is administered. The backup D-channel may not come into service in time to handle the restoration attempt.

- Set Time Command

  When you change the system time via the set time command, all scheduled AC are examined. If the time change causes an active AC to be outside its scheduled period, the AC drops. If the time change causes an inactive AC to be within its scheduled period, the switch attempts to establish the AC.

  If any AC (scheduled or continuous) is in retry mode and the system time changes, the switch attempts to establish the AC.

- System Measurements

  Access endpoints are not measured. All other trunks in an AC are measured as usual.

- Terminal Dialing

  Turn off terminal dialing for data modules involved in an AC. This prevents display of call-processing messages (INCOMING CALL,...) on the terminal.

- Trunk Groups

  To invoke auto restoration, route an AC over SDDN trunks. Because a successful restoration depends on a SDDN path, keep some SDDN trunks idle.

Administration Change Notification

You can use Administration Change Notification to notify adjunct systems when MultiVantage administration data is changed. This is intended to keep a client application running on an adjunct, such as Enterprise Directory Gateway, in sync with changes in the switch.

You can request notification of any changes to MultiVantage administration by entering the command **notify history** from the client application. MultiVantage will continue to send notification to the adjunct over the Operations Support System Interface (OSSI) link until the command is cancelled.

Detailed description

Administration Change Notification tracks changes made via the SAT, INADS port, Property Management System, Call Management System, Avaya Site Administration, Avaya Network Administration, or Avaya Directory Gateway. It also tracks any changes made through a telephone interface, such as TTI (Terminal Translation Initialization), Personal Station Access, and Terminal Self Administration.

The MultiVantage System only notifies the adjunct that a data object has been changed, but does not provide details of the change. In order to obtain these details, the adjunct must request this information from the switch over a separate link.

Administrable Language Displays

You can display messages in English, French, Italian, Spanish, or a user-defined language on attendant consoles and phones that have 40-character displays.

Detailed description

You can select a language for messages that appear on phones and attendant consoles that have 40-character displays. You can choose one of five languages: English, French, Italian, Spanish, or “user-defined.” The meanings of the messages do not change, only the language.

If your company uses 32-character display phones, you cannot choose a display language. These phones, including the hybrid MERLIN 7315H and 7317H phones, default to English.

Related topics

You can display the following types of information in English, French, Italian and Spanish:

- Automatic Wakeup
- ASAI
- Busy Verification of Terminals and Trunks
- Call Appearance buttons
- Call Detail Recording
- Call Progress Feedback Displays
- Class of Restriction
- Date-Time Mode - Time Not Available
- Days of the Week
- Months of the Year
- Do Not Disturb
- Enhanced Abbreviated Dialing
- Integrated Directory
- ISDN
- Leave Word Calling
- Malicious Call Trace
- Emergency Access to Attendant
- Queue Status

- Miscellaneous Call Identifiers
- Party Identifiers
- Property Management Interface
- Security Violation Notification
- Stored numbers
- Station hunting
- Time-of-day Routing
- Transfer messages

Alternate facility restriction levels

Alternate Facility Restriction Levels (AFRL) allows a second set of facility restriction levels within a route pattern or for lines or trunks. Attendants and system administrators can activate the alternate FRLs and change users' access to lines and trunks. For example, a company can use AFRL to disable long-distance calling at night to prevent unauthorized staff from making long-distance calls.

AFRL alters the route patterns for originating phones, originating trunks, and dialed authorization codes. If AFRL is active, Travelling Class Mark (TCM) is also set to a new FRL value and the TCM information recorded in the billing data (CDR) is the AFRL value, not the original TCM.

CAUTION:

AFRL impacts Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) call routing because it may change routing preferences. Using AFRL on tandem and tie-trunk applications affects entire networks. Calls that are part of a cross-country private network and may have to be routed further may be blocked.

Detailed description

You can administer an ALT-FRL button to any attendant console or any station to activate and deactivate the AFRL. Pressing the ALT-FRL button may affect the status of other buttons.

When AFRL is active, the user may notice a change in calling privileges. Consider notifying users of the changes, and prepare your telecommunications department to handle inquiries.

Authorization codes

Authorization codes prevent unauthorized access to various facilities. When a user dials an authorization code, your system checks the code. If it is not valid, the call is intercepted. If the code is valid, the system determines an associated COR and FRL. If AFRL is activated, the AFRL level is used.

For example, a user whose FRL is 1 attempts a long-distance call. AFRL is active and maps to AFRL 3. The desired trunk has an FRL of 7, and the call is blocked. In the example below, an Authorization Code set to1234567 has a COR of 3 with an FRL 5, which is still not high enough to access the desired trunk. However, AFRL is active and FRL 5 maps to FRL 7. The call is allowed.

Authorization code to COR Table

Authorization Code	COR
1234567	3
1234568	2
1234569	3
2222222	3

COR to FRL Table

COR	FRL
1	1
2	3
3	5
4	7

FRL to AFRL Table

FRL	AFRL
0	3
1	3
2	3
3	3
4	7
5	7
6	7
7	7

Originator COR is 1
Trunk Desired COR is 4

Figure 56. Example of authorization codes with AFRLs

Analog busy automatic callback without flash

This is an automatic callback for analog stations without flashing the hook. It will be applied only when the called station is busy and has no other coverage path or call forwarding. The caller can enable the automatic callback without flashing the hook or entering the feature access code.

Detailed description

With Analog Busy Automatic Callback without Flash, when the caller places a call through an analog station, and the called station is busy and has no coverage path or forwarding, the caller hears an announcement and is presented with options. Depending on the caller’s selection, the call is queued to Automatic Callback, routed to an extension, or dropped.

Configurations

This feature will be available on the following platforms:

Avaya S8100 Media Server with an Avaya G600 Media Gateway

Avaya S8100 Media Server with an Avaya CMC1 Media Gateway

Avaya MultiVantage on a DEFINITY Server CSI

Avaya DEFINITY Server SI with an Avaya SCC1 Media Gateway

Avaya DEFINITY with an Avaya SCC1 Media Gateway

Related topics

‘‘Automatic Callback’’ on page 1370

Announcements

You can record announcements for people to hear when they call in to your office. For example, you can let callers know that their call cannot be completed as dialed, that their call is in queue, or that all the lines are busy.

Announcements can be integrated or external. Integrated announcements reside on a circuit pack in the switch carrier. Your system can store multiple announcements on each circuit pack up to the system capacity.

For MultiVantage software release R11 or later, multiple telephone sessions are allowed, with one session associated with each active integrated announcement circuit pack.

External announcements are stored and played back from adjunct equipment. For more information on external announcements, see *Avaya MultiVantage™ Call Center Software Release 11 Guide to ACD Call Centers.*

Any announcement stored on a circuit pack can play through any port on the circuit pack. Any announcement (not administered for "barge-in") can play simultaneously through multiple ports. All ports can play the same announcement at the same time, and the system can connect multiple users to each of these announcements.

Three types of announcements are:

- delay announcement — explains the reason for the delay and encourages caller to wait
- forced announcement — explains an emergency or service problem. Use when you anticipate a large number of calls about a specific issue.
- information announcement — gives the caller instructions on how to proceed, information about the number called, or information that the caller wants

TN2501AP announcements

The TN2501AP is an integrated announcement circuit pack that allows you to store and manage announcements over a Local Area Network (LAN). For this reason, it is sometimes referred to as the Voice Announcements over the LAN (VAL) circuit pack.

The TN2501AP can make use of a separate announcement management tool, VAL Manager. VAL Manager provides a PC interface from which you can add, change, delete, save, and restore announcements. For more information, see ''VAL Manager'' on page 456.

Despite the name, announcements are not actually played over the LAN, but can be transferred to and from the TN2501AP circuit pack over the LAN.

The TN2501AP

- is available with both Offer Category A (ECS and ProLogix Solutions) and Category B (BCS and GuestWorks).
- works with r, si, csi, DEFINITY One, Avaya IP600 platforms.
- can be updated with firmware files downloaded directly over the LAN. The files are downloaded through the TN2501AP's 10/100Mb ethernet interface, not through the TN799 C-LAN circuit pack.
- has a 10/100MB ethernet LAN connection through the Backplane Adapter.
- has up to 31 ports for announcement playback, with an additional port for recording announcements directly to the TN2501AP circuit pack through a system telephone.

You cannot save or restore announcements to a TN2501AP circuit pack to/from

- a TN750C circuit pack
- flash cards
- tape
- magneto optical disks

Administration options for TN2501AP announcements

If you use the TN2501AP circuit pack, you can manage announcements through

- the System Access Terminal (SAT).
- Avaya Site Administration
- a PC running the VAL Manager application. VAL Manager is the easiest way for you to manage your TN2501AP announcements. For more information, see your Avaya representative.

You can record VAL announcements

- from a system phone.
- on a computer or any device that supports recording .wav files in a VAL-compatible format (CCITT µ-Law or A-Law, 8KHz, 8-bit mono).
- at a professional recording studio.

  See ''Announcement file format requirements'' on page 445 for more information about recording parameters.

The following table shows the VAL administration tasks and which interface you can use to complete each task. The TN2501AP circuit pack was designed with your efficiency and flexibility in mind, so try combining methods to accommodate your particular installation. See ''Getting started with the TN2501AP'' on page 443 and ''Managing VAL Announcements Using FTP'' on page 449 for instructions for performing these tasks.

Table 23. VAL administration tasks

Tasks	Methods		
	SAT	**FTP**	**VAL Manager**
Adding announcement extensions	X		X
Deleting VAL announcements	X	X	X
Saving or backing up announcements		X	X
Restoring an announcement		X	X
Moving announcement files or administration	X	X	X
Recording announcements	Professional or computer recordings or system phone		

System restarts - TN750

When someone powers up the system or inserts or resets an announcement circuit pack, the system checks the circuit pack for announcements. If the system finds no announcements on the circuit pack, and there are recorded announcements stored in system memory, the system restores the announcements from system memory to the announcement circuit pack.

CAUTION:

The announcements that are automatically restored are the last announcements saved to system memory. If multiple circuit packs are used, system memory might not contain the announcement for the B or A circuit pack.

The system automatically restores announcements to only one announcement circuit pack. The system does not restore to announcement circuit packs that have built-in memory.

About barge-in

You can allow callers to begin listening to an announcement after the system has begun playing the message. This is called “barge-in.” Use barge-in with auxiliary trunk announcements, DS1 announcements, and integrated announcements.

With barge-in, only one port plays the announcement at any one time. The system routes a call to the announcement, immediately connects the call to the port, and the caller hears the announcement as it is playing. You can set up barge-in announcements to repeat continually while callers are connected to the port. The caller listens until the system plays the entire announcement.

Interactions

- Automatic Wakeup

  Recorded Announcement allows Automatic Wakeup to use the built-in announcement circuit pack in place of an Audichron adjunct.

  If you use an integrated, multiple-integrated, or external type of announcement for Automatic Wakeup, you can also administer the announcement to repeat and to allow “barge-in” as a queue type. The benefit of repeating announcements and “barge-in” queues is that you do not need a separate port for each wakeup announcement. When guests pick up an announcement at a particular time, they use only one port and the message repeats until the last guest hangs up and the message ends.

Related topics

Refer to ‘‘Managing announcements’’ on page 431 for information on how to administer announcements on your switch.

Refer to the *Avaya MultiVantage™ Hardware Solutions Guide* for more information on the TN2501AP and the TN750-series announcement circuit packs.

Refer to the *Made Easy Tool for DEFINITY® Media Server Configurations* for information related to installing and upgrading a system with the TN2501AP.

Refer to ‘‘Managing vectors and VDNs’’ on page 228 for information on how to play an announcement for a call in a queue.

Refer to ‘‘Announcement Type’’ on page 834 for information on setting up announcements for hotel guests.

Answer detection

For proper call detail recording and accurate billing, you need to know when your outgoing calls are answered. Answer detection means the method a switch uses to determine whether an outgoing call has been answered.

Brief description

MultiVantage provides 3 ways to determine whether the called party has answered a call.

Call classification

A call-classifier circuit pack detects tones and voice-frequency signals on the line to determine whether a call has been answered. This method is fairly accurate: calls that are answered normally are usually classified correctly. But there are exceptions:

- Miscellaneous tones, such as confirmation tones, may be classified as answers.
- Loud background noise may activate answer detection, causing a call to be classified as answered even if the call is not connected.
- Some calls that are answered may be incorrectly classified as fast busy signals.
- Call classifier circuit packs also don’t recognize Special Information Tones (SIT) as answers.

Of course, the system generates a call record for any call that is classified as answered whether the classification is correct or not. If Call Classification incorrectly classifies a call as answered, and then the call is subsequently answered, the call duration reported by CDR includes the both time between the incorrect classification and the actual answer and the remaining duration of the call.

If you want to use call classification, on the System Parameter Customer-Options screen verify the Answer Supervision by Call Classifier field is **y**. If not, contact your Avaya representative. You also must have a call classifier circuit pack of the correct type. To find out what circuit packs you need, refer to *Avaya MultiVantage™ Hardware Solutions Guide.*

Network answer supervision

The central office sends a signal to the originating switch to indicate that the far end has answered. If a call traveled over a private network before reaching the central office, the signal is transmitted back over the private network to the originating switch. This method is extremely accurate, but it is not available over most loop-start trunks (for example, CO, FX and WATS trunks in the US).

Network answer supervision does not override answer supervision by timeout.

Answer supervision by timeout

You set a timer for each trunk group, using the Answer Supervision Timeout field on the Trunk Group screen. Or you set a circuit pack timer for the ports on that circuit pack, using the Outgoing End of Dial (sec) field on the Trunk Group screen. If the caller is off-hook when the timer expires, the system assumes that the outgoing call has been answered. This is the least accurate method. Calls that are shorter than the timer duration do not generate call records, and calls that ring for a long time produce call records whether they are answered.

If network answer supervision is received, it overrides answer supervision by timeout.

Interactions

- Call Detail Recording

  Answer Detection provides more accurate call records where tone detection is possible and Network Answer Supervision is not received.

- Call Prompting

  Call classification competes with Call Prompting for ports on the call classifier circuit pack.

- CallVisor ASAI

  Call classification competes with CallVisor ASAI switch-classified calls for ports on the call classifier circuit pack. Answer Detection triggers reporting of a connect event to ASAI.

Related topics

Refer to ‘‘Administering answer detection’’ on page 428 for instructions.

Refer to ‘‘Trunk Group’’ on page 1167 for definitions of the fields used to administer answer detection.

Attendant Features

This section describes the following MultiVantage features that you may want to administer for your console operators:

Each feature indicates how to administer the Attendant Console screen, Console Parameters screen, and other system-wide screens to enable the feature.

Attendant Call Waiting

Attendant Call Waiting allows an attendant-originated or attendant-extended call to a busy single-line telephone to wait at the called telephone so that the attendant can handle other calls.

If you want the attendant to be able to send calls to busy single-line phones, set the Att. Call Waiting Indication field to Y on the Station screen for each single-line phone.

When the single-line phone receives a waiting call, the phone user hears a call-waiting signal. You can administer the number of bursts (1, 2, or 3) in the call-waiting signal by changing the Attendant Originated Calls field on the Feature-Related System Parameters screen.

If the attendant activates Attendant Call Waiting, and the Timed Reminder on Hold interval or the Return Call Timeout interval expires without the call being answered, the call returns to the console. You can modify these intervals on the Console Parameters screen.

Interactions

- Automatic Callback

  Activating Automatic Callback at the called telephone denies Attendant Call Waiting.

- Call Coverage

  Attendant Call Waiting calls redirect to coverage if the called phone has Data Privacy or Data Restriction activated. If one of these conditions exists, and you assign call coverage to a telephone, and the user activates Send All Calls or coverage criteria is met, the call redirects to coverage.

  — The Coverage Don’t Answer interval specifies how long a call remains directed to the called telephone before redirecting to coverage. Attendant Call Waiting if applicable on the call, is active for the duration of the Don’t Answer interval only. At the end of this interval, the call redirects to coverage.

  — If the Return Call Timeout (Timed Reminder) interval expires before the Don’t Answer interval expires, the call does not go to coverage, but returns to an attendant console. If the Don’t Answer interval expires first, the call redirects to coverage, but can still return to the console if a coverage point does not answer the call before the Return Call Timeout.

  — If the Station Hunting field is assigned and the called telephone is busy, the call redirects to the Hunt To Station Assignment.

- Data Privacy, Data Restriction

  Activating Data Privacy or Data Restriction at the called telephone denies Attendant Call Waiting.

- DDC and UCD

  Calls to a DDC or UCD group do not wait. However, they can enter the group queue, if provided.

- Loudspeaker Paging Access

  Activating Loudspeaker Paging Access at the called telephone denies Attendant Call Waiting.

- Music-on-Hold Access

  The calling party hears music if the call is a trunk-transferred call administered to receive Music-on-Hold. Otherwise, the calling party hears ringing.

- Recorded Telephone Dictation Access

  Activating Recorded Telephone Dictation Access at the called telephone denies Attendant Call Waiting.

Attendant Control of Trunk Group Access

Attendant Control of Trunk Group Access allows the attendant to control trunk groups, and prevents telephone users from directly accessing a controlled trunk group. The attendant gains direct access to an outgoing trunk group by pressing the button assigned to that trunk group.

Each attendant console has 12 designated Trunk Hundreds Select buttons that can be administered for Attendant Control of Trunk Group Access. You can also administer each console with up to 12 feature buttons for Trunk Hundreds Select buttons, which gives you up to a total of 24 buttons.

Each Trunk Hundreds Select button has busy lamps that light when all the members of the associated trunk group are busy. If you administer one of the 2-lamp feature buttons on a basic console as a Trunk Hundreds Select button, use the bottom lamp as the busy lamp. These buttons have 2 additional lamps for Attendant Control of Trunk Group Access. The 2 lamps are:

- Warn (warning) lamp

  Lights when the administered number of trunks are busy in the associated trunk group. You administer the Busy Threshold field on the associated Trunk Group screen to determine when to light this warning lamp.

- Cont (control) lamp

  Lights when the attendant activates Attendant Control of Trunk Group Access for the associated trunk group. Assign act-tr-grp and deact-tr-g buttons on the Attendant Console screen to allow the attendant to activate and deactivate control of the trunk group access.

Interactions

- Authorization Codes

  Authorization codes do not collect when a trunk group has an incoming destination set to the attendant.

- Automatic Route Selection and Automatic Alternate Routing (ARS/AAR)

  Activating Attendant Control of Trunk Group Access removes the controlled trunk groups from the ARS and AAR patterns. Deactivating the feature reinserts the groups into the patterns. ARS calls do not route to the attendant.

- QSIG

  QSIG trunks do not support Attendant Control of Trunk Group Access.

- Uniform Dial Plan

  Activating Attendant Control of Trunk Group Access removes the controlled trunk groups from preferences. Deactivating the feature enables the UDP to access the trunk groups.

Attendant Direct Extension Selection

Attendant Direct Extension Selection (DXS) with busy lamp field allows the attendant to track extension status (idle or busy) and to place or extend calls to extension numbers without having to dial the extension.

Standard DXS Tracking

The basic selector console has 8 Hundreds Select buttons and 100 DXS buttons. The enhanced selector console has 20 Hundreds Select buttons and 100 DXS buttons. You can assign 12 additional Hundreds Select buttons to feature buttons on the attendant console.

However, as you assign these feature buttons, note that the total number of Hundreds Select buttons per attendant (including both attendant-console feature buttons and selector-console buttons) cannot exceed 20.

Enhanced DXS Tracking

Enhanced DXS Tracking can help you if you have more than 100 telephones, but you use a console that does not have Hundreds Select buttons administered. It can also help if you have more telephones than you do Hundreds Select buttons (and thus have hundreds groups that are administered with Hundreds Select buttons).

To use Enhanced DXS, assign a Group Select button on the Attendant Console screen. This button allows the attendant to track and extend calls to telephones that do not have associated Hundreds Select buttons. You can not use Enhanced DXS Tracking if your extensions have fewer than three digits.

Group Display button

You can administer a Group Display button on the Attendant Console screen to help the attendant track extension status. When the attendant presses this button, the system displays the range of extensions currently tracked by the selector console. Administer the Group Display button for either the feature area or the display area of the console.

If the attendant selects this button, the system identifies the digits associated with a Hundreds Select button — unless it finds no Hundreds Select button is lit, in which case it identifies the digits last entered with the Group Select button. The system continues to track the selected group of extensions until the attendant selects a new group of extensions.

Attendant Intrusion

The attendant intrusion (Call Offer) button allows an attendant to intrude on an existing call to offer a new call or message to the intruded party.

When the attendant releases the intruded call, the source party waits at the intruded party's analog telephone or holds on an available line appearance on a digital telephone.

Interactions

- Intrusion is denied in the following cases:
  - A telephone is on a conference call with administered maximum number of conferees
  - A call is established with Data Privacy activated
  - Establish a call with Data Restriction activated
  - A telephone is a forward-to point of another telephone
  - A telephone is busy talking to another attendant
- If a call is already call waiting for the intruded party, the source (split from attendant) party cannot wait for the intruded party using Call Waiting.
- The attendant display shows the character '1 wait' or '1 busy' if an intrusion is possible. Otherwise, the display shows 'wait' or 'busy'.
- The system provides Attendant Intrusion on remote telephones via TGU/TGE trunks (Italy only).

Attendant Override of Diversion Features

Attendant Override of Diversion Features (override button) allows an attendant to bypass call-diversion features activated by a called extension. A diversion feature is any feature that, when activated, causes a call to redirect from the called telephone. Send All Calls, Call Coverage, and Call Forwarding are diversion features.

You should explain to your attendants that they can use this feature with the Attendant Intrusion to place an emergency or urgent call to a telephone user.

Attendant Serial Calling

Attendant Serial Calling enables the attendant to transfer trunk calls that return to the same attendant after the called party hangs up. Once outside callers reach an attendant, they can use the same line into the switch for multiple calls. Attendant Serial Calling is useful if trunks are scarce and Direct Inward Dialing services are unavailable.

To allow your attendant to use serial calling, assign a serial-cal button on the Attendant Console screen. The Attendant Serial Calling feature is valid only on calls that have only one trunk on the connection.

You can define a priority queue for Serial Calls on the Console Parameters screen.

Interactions

- Centralized Attendant Services

  Attendant Serial Calling does not work with Centralized Attendant Services.

- DCS

  Attendant Serial Calling works in a DCS environment only if the attendant activates it on the same node as the trunk to which the attendant is connected. Do not conference the incoming trunk call with a DCS party when activating. This would put two trunks on the connection.

Attendant Vectoring

Attendant Vectoring allows you to establish an attendant vector directory number (VDN) and send attendant group calls through vector processing. This is useful when you want more flexibility with how calls are routed when the system is in Night Service mode. For more information, see *Avaya MultiVantage Call Vectoring/Expert Agent Selection (EAS) Guide*.

Auto Start and Don’t Split

Auto Start allows the attendant to initiate a call by pressing any key on the keypad without having to first press the Start button.

If an attendant enables Auto Start and dials an AAR number where the min and max in the AAR analysis table are not equal, the attendant must dial a **#** after the digit string or the call cannot process.

You can assign a dont-split button on the Attendant Console screen which allows attendants to deactivate Auto Start. To deactivate auto start, the attendant presses the Don’t Split button. When Don’t Split is active, keys pressed on the keypad are heard by the parties on the call.

To reactivate Auto Start, and allow end-to-end signaling, the attendant again presses the Don’t Split button, presses Cancel, or lets the current call terminate.

Interactions

- CDR — Account Code Dialing

  If the system is using Call Detail Recording Account Code Dialing, Auto Start and Don’t Split is not activated.

- Visually Impaired Attendant Service

  If VIAS is activated or deactivated while Don’t Split is active, Don’t Split deactivates.

Attendant Timers

Attendant timers automatically alert the attendant after an administered time interval. The attendant can reenter the call and decide whether to terminate the call or permit the waiting to continue. You administer the timers on the Console Parameters screen.

Attendant Timers include:

- Unanswered DID Call Timer — Specifies how long a DID call can go unanswered before it routes to the administered DID/TIE/ISDN Intercept Treatment.
- Attendant Return Call Timer — For unanswered calls that were extended by the attendant, they are returned to the same attendant who released them if the attendant is available. Otherwise they return to the attendant-group queue. The Attendant Return Call Timer is not set for calls extended from one attendant to another individual attendant. A transferred call that times out redirects to an attendant after an interval equal to the Attendant Return Call timer.
- Attendant Timed Reminder of Held Call Timer — Specifies how long a call is held. When the timer expires, the held call alerts the attendant. The message `hc` appears on the attendant display. You can administer either a high-pitched ring or a primary alert.

- Attendant No-Answer Timer — Specifies how long a call that terminates at an attendant console can ring with primary alerting. When the call reaches this interval setting, it rings with a secondary, higher-pitch ring. A disabled Attendant No Answer Timer's ringing pattern does not change over from the primary to the secondary pattern. If the call remains unanswered during this interval, it routes to the attendant group and console where the call was placed in a Position Busy state. This feature does not apply to calls placed to the attendant's extension or to calls originated by the attendant.
- Attendant Alerting Interval (Timed Reminder) — Specifies how long a call that terminates at an attendant console can ring with secondary alerting. When the call reaches this interval, the attendant console is placed into position busy mode and the call forwards to the attendant group. If the console where the alerting interval is reached is the last active day console, then the system goes into night service if night service is enabled. This feature does not apply to calls placed to the attendant's extension or to calls originated by the attendant.

  You can disable the alerting interval. In this case, a call continues to ring at the original attendant's extension until the caller hangs up or another feature disconnects the call (for example, reaching the timeout limit for unanswered DID calls during night service.)
- Line Intercept Tone Timer — Specifies how long line intercept can be. For example: LITT:10 seconds means that line intercept stops after 10 seconds.

Interactions

- Call Coverage

  If a telephone user transfers a call to an on-premises telephone and the call remains unanswered at the expiration of the Timed Reminder Interval, the call redirects to an attendant. Redirection occurs even if the call redirects via Call Coverage or Call Forwarding from the transferred-to telephone.

  An attendant-extended call redirects to coverage instead of returning to an attendant if the coverage criteria are met before the Timed Reminder Interval expires. However, unanswered calls return to an attendant at the expiration of the interval.

  If a call alerts an attendant as a coverage call (unanswered station-to-station call with the "attd" (attendant) in the called telephone's coverage path screen), the secondary alerting tone does not sound.
- Centralized Attendant Service

  If an attendant at the main location transfers a call from a branch location to an extension at the main location, the timed reminder does not apply and the call does not return to the attendant if unanswered.

Visually Impaired Attendant Service

Visually Impaired Attendant Service (VIAS) allows a visually-impaired attendant to listen to an audio description of each feature button in Inspect mode. It provides the description in either British English or Italian.

The attendant presses the Inspect mode to locate each button and then presses a feature button to determine the feature assigned to the button without actually executing the feature.

The six VIAS attendant buttons are:

- Visually Impaired Service Activation/Deactivation button: activates or deactivates the feature. All ringers previously disabled (for example, recall and incoming calls) become reenabled.
- Console Status button: voices whether the console is in Position Available or Position Busy state, whether the console is a night console, the status of the attendant queue, and the status of system alarms.
- Display Status button: voices what is shown on the console display. VIAS support is not available for all display features (for example, class-of-restriction information, personal names, and some call purposes).
- Last Operation button: voices the last operation performed.
- Last Voiced Message button: repeats the last voiced message.
- Direct Trunk Group Selection Status button: voices the status of an attendant-monitored trunk group.

Some changes on the attendant console are automatically voiced (for example, alarms reported, night service activated, and call thresholds reached).

Audible Message Waiting

Audible Message Waiting places a stutter at the beginning of a station dial tone on a station that has a message waiting. Audible Message Waiting is particularly useful for visually impaired people who may not be able to see a message light.

Messages for a station can be waiting in system memory (to be accessed via display or voice synthesizer), Property Management System (PMS) or AUDIX. When the system loses synchronization between telephones and message-status data, use Clear Message Waiting Indicators to turn off message-waiting indicators.

You typically assign Audible Message Waiting on phones without message-waiting lights, such as analog telephones.

Audible Message Waiting requires a separate software right-to-use fee. Audible Message Waiting may not be applicable in countries that restrict the characteristics of dial tones provided to users.

Related topics

Refer to ''System-Parameters Customer-Options'' on page 1121 for information about and fields descriptions on the System Parameters Customer-Options screen. Complete the Audible Message Waiting field on this screen to administer audible message waiting.

Refer to ''Station'' on page 1066 for information about and fields descriptions on the Station screen. Complete the Audible Message Waiting field on this screen to administer audible message waiting.

Authorization codes

Authorization codes provide the means for extending control of system users' calling privileges. They extend calling-privilege control and enhance security for remote-access callers.

NOTE:
To maintain system security, Avaya recommends you use authorization codes.

Authorization codes may be used to:

- Override a facility restriction level (FRL) assigned to an originating station or trunk
- Restrict individual incoming tie trunks and remote-access trunks
- Track Call Detail Recording calls for cost-allocation purposes
- Provide additional security control

You can make authorization codes mandatory by setting, on the trunk group screen, the Auth Code field to **y**. Refer to ''Trunk Group'' on page 1167 for more information.

More information

When you dial an authorization code, the FRL assigned to the extension number, attendant console, incoming trunk group, or remote access trunk group being used for the call is replaced by the FRL assigned to the authorization code. The new FRL functions the same as the one it replaces; however, the new FRL may represent greater or lesser calling privileges than the FRL that it replaces. Access to any given facility depends on the restrictions associated with the authorization code FRL.

Example

A supervisor is at a desk of an employee and wants to make a call that is not normally allowed by the FRL assigned to that employee's extension. The supervisor, however, can still make the call by dialing an authorization code that is assigned an FRL that is not restricted from making that type of call.

For security reasons, authorization codes range from 4 to 13 digits. The number of digits in the codes must be a fixed length for a particular MultiVantage system.

NOTE:
Once established, the number of digits (4 to 13) in the authorization code remains fixed unless all codes are removed and re-entered. All authorization codes used in the system must be the same length.

Incoming trunk groups within a system may be administered to always require an authorization code. The system applies recall dial tone to a call when the user must dial an authorization code. If the user dials a correct authorization code within 10 seconds (interdigit timeout), the call completes as dialed. If the user does not dial an authorization code or dials an incorrect authorization code, the call routes to the attendant, or routes to intercept tone, depending on system administration.

Normally, Direct Inward Dialing (DID) trunks should not require authorization codes. However, it can be done and care should be taken when administering DID trunks to require an authorization code, because different type calls could terminate at different endpoints, and requiring an authorization code could be confusing to the caller.

A Cancellation of Authorization Code Request (CACR) digit may be administered. The CACR digit cancels the 10-second interval between dialing. When the CACR digit is dialed, the call immediately routes according to system administration. (Incoming trunk calls receive intercept treatment or go to the attendant.) Other calls receive intercept treatment unless the user's FRL is high enough to route the call. A CACR digit from an off-premises extension over DID/Tie trunks use DID/Tie trunk intercept treatment. Internal calls receive intercept tone.

CAUTION:

Do not program passwords or authorization codes onto auto dial buttons. Display telephones display the programmed buttons, providing internal abusers access to the auto dial buttons to originate unauthorized calls. If passwords or authorization codes must be programmed onto auto dial buttons, use the ~s (suppress) character to prevent displaying the codes.

For more information, refer to *Avaya Products Security Handbook*.

AAR and ARS Calls

Each authorization code is assigned a Class of Restriction (COR) that contains an associated FRL. Within a system, access privileges are determined by the FRL assigned to the facility where the call is originated. When an Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) call is dialed, the system allows or denies the call based on the FRL of the originating station. COR is used to restrict internal or non-AAR/ARS calls.

Authorization codes are given to individual users and provide a method of specifying the level of calling privileges for that user regardless of the originating facility. Once an authorization code is required and dialed on an AAR/ARS call, the FRL assigned to the authorization code replaces the originating FRL and controls and defines the user's privileges.

An AAR or ARS call originated by a system user or routed over an incoming tie trunk may require a dialed authorization code to continue routing.

Extreme care should be taken when administering authorization codes, so that a user does not have to dial the authorization code more than once. For example, if a user makes an AAR or ARS call and the user's FRL is not high enough to access any of the trunks in the routing pattern, the system prompts the user for an authorization code. If the FRL assigned to the authorization code is high enough to access the next trunk group in the routing pattern, the user is not prompted to dial the code again. If the call is routed through another switch, the user may be required to dial an authorization code again. This type of situation can be avoided through careful administration.

When an authorization code is required on some, but not all, trunk groups, the system prompts for an authorization code when the originating FRL is not adequate to access the next available trunk group in the routing pattern.

Considerations

- From remote locations users typically access authorization from touch-tone stations. However they can also do so from rotary dialing stations at specified authorization-code-forced locations that follow appropriate trunk administration practices. Rotary station users access attendants via Listed Directory Numbers (LDN) or remote access numbers and can experience a 10-second timeout.
- The use of Authorization Codes does not limit other call-control methods such as Toll Restriction, Miscellaneous Trunk Restriction, and Outward Restriction.
- For security reasons, do not assign authorization codes in sequential order. Assign random number barrier codes and authorization codes to users so if a hacker deciphers one code, it will not lead to the next code.
- If timeout to attendant does not occur or CACR digit codes are dialed instead of authorization codes, the system assumes that invalid authorization codes were dialed and the caller is given intercept tones.
- Authorization codes impact calling privileges by:
  - Changing an outgoing-call FRL when it is insufficient to access preferred routing patterns assigned by AAR/ARS. An FRL is assigned to a COR associated with user authorization codes. No additional COR data is assigned.
  - Overriding COR for remote access calls assigned to barrier codes, when required. For remote-access calls, if an authorization code is required, the user is assigned the COR of the dialed authorization code, with all connected data, such as the FRL. This COR overrides the COR assigned to any required barrier code.
- Incoming trunk calls that require authorization codes do not change user privileges.

Interactions

- AAR/ARS Partitioning

  Partitioned group numbers are assigned by COR and Authorization Codes can change Classes of Restriction. Therefore, Partitioned Group Numbers can be changed on incoming remote access calls by authorization codes. For originating calls, user Classes of Restrictions determine Partitioned Group Numbers.

- Cancellation of Authorization Code Request

If	Then
CACR =1	■ Authorization ≠ 1
Network = DEFG1, DEFG3 or DEF ECSR5	■ CACR can be #
Network - S85s, DIM switch	■ CACR = 1 (default)

- COR and FRL

  Authorization codes used for AAR/ARS calls override associated FRL.

  Associated Classes of Restriction determine remote-access user privileges.

- Forced Entry of Account Codes and Call Detail Recording

  For 94A LSU (no longer supported) and 3B2 CDRU (no longer supported) 18-word records, authorization codes are output if administered account-code lengths are fewer than six digits. For 59-character records, authorization codes are never recorded.

  Authorization codes are recorded after destination addresses are dialed. Invalidly-dialed authorization codes are recorded, and patterns can be traced using CDR printouts.

Related topics

Refer to ''Barrier codes'' on page 1388 to use barrier codes to prevent unauthorized system access.

Refer to ''Station'' on page 1066 for setting a class of restriction on a station phone.

Refer to ''Class of Restriction'' on page 636 to set a facility restriction level on a phone.

Refer to ''Route Pattern'' on page 1039 for information concerning the sequence of trunk groups in which an attempt is made to route a call.

Refer to ‘‘QSIG to DCS TSC Gateway screen’’ on page 1029 to permit authorized callers from remote locations to access your system.

Refer to ‘‘Trunk Group’’ on page 1167 to require an authorization code be dialed to complete incoming calls on a trunk group.

Refer to ‘‘Setting up authorization codes’’ on page 398 for instructions.

Automated Attendant

Automated Attendant uses vector commands to allow a caller to enter the extension of the party that he or she would like to reach. The call is routed by the vector to that extension.

Refer to *Avaya MultiVantage™ Call Center Software Release 11 Call Vectoring and Expert Agent Selection (EAS) Guide* for a detailed description of Automated Attendant and for a sample vector that can be used for Automated Attendant. The guide contains information that is critical to the effective and efficient use of Automated Attendant.

You can administer any display-equipped phone or attendant console with a Caller Information CALLR-INFO button. The button displays digits collected for the last **collect digits** command.

Automated Attendant competes with several features for ports on the call classifier — detector circuit pack or equivalent.

Interactions

- AUDIX

  Automated Attendant gives the caller the option of leaving a message or waiting in queue for an attendant. Refer to “Message Collection” in Chapter 5 of the *Avaya MultiVantage™ Call Center Software Release 11 Call Vectoring and Expert Agent Selection (EAS) Guide*.

- Authorization Codes

  If authorization codes are enabled, and a *route-to* command in a prompting vector accesses AAR or ARS, if the VDN’s FRL does not have the permission to use the chosen routing preference, then the system does not prompt for an authorization code and the *route-to* command fails.

- CallVisor ASAI

  ASAI-provided digits can be collected by the Call Vectoring feature via the *collect* vector command as dial-ahead digits. CINFO is passed to CallVisor ASAI.

- Hold

  If a call is put on hold during the processing of a *collect* command, the command restarts, beginning with the announcement prompt, when the call is taken off hold. All dialed-ahead digits are lost. Similarly, if a call to a vector is put on hold, vector processing is suspended when a *collect* command is encountered. When the call becomes active, the *collect* command resumes.

- Inbound Call Management (ICM)

  You can use Automated Attendant to collect information that may later be used by an adjunct to handle a call.

- Transfer

  If a call to a VDN is transferred during a *collect* command, the *collect* command restarts when the transfer is complete, and all dialed-ahead digits are lost. Similarly, if a call to a vector is transferred, vector processing is suspended when a *collect* command is encountered. When the transfer is complete, the *collect* command resumes. Attendant extended calls do suspend vector processing in the same way as transferred calls.

Related topics

Refer to ‘‘System-Parameters Customer-Options’’ on page 1121 for information about creating an automated attendant.

Refer to ‘‘Feature-Related System Parameters’’ on page 766 for information on the number of seconds before the Collect Digits command times out.

Refer to ‘‘Vector Directory Number’’ on page 1235 for information about and field descriptions on the Vector Directory Number screen.

Refer to ‘‘Announcements/Audio Sources’’ on page 588 for information on providing an automated attendant announcement.

Refer to ‘‘Call Vector’’ on page 616 for information on completing a new screen for each automated attendant vector.

Refer to ‘‘Hunt Group’’ on page 840 for information on whether a hunt group will be vector controlled.

Refer to ‘‘Station’’ on page 1066 for information on button assignments.

Refer to ‘‘Attendant Console’’ on page 595 for information on button assignments.

Automatic Callback

Automatic Callback allows internal users who placed a call to a busy or unanswered internal telephone to be called back automatically when the called telephone becomes available.

When a user activates automatic callback, the system monitors the called telephone. When the called telephone becomes available to receive a call, the system originates the automatic callback call. The originating party receives priority ringing. The calling party then lifts the handset and the called party receives the same ringing provided on the original call.

A single-line telephone user activates this feature by pressing the Recall button or flashing the switchhook and then dialing the automatic callback access code. A single-line user can activate automatic callback for only one call at a time.

A multi-appearance telephone user can activate automatic callback for the number of automatic callback buttons assigned to the telephone. After placing a call to a telephone that is busy or that is not answered, the caller simply presses an idle automatic callback button and hangs up.

If the calling telephone user answers an automatic callback call, and for some reason the called extension cannot accept a new call, the calling user hears confirmation tone and then silence. The call is still queued.

Users cannot activate automatic callback for calls to:

- A telephone assigned Termination Restriction
- An extension with automatic callback already activated toward it
- A data terminal (or data module)
- An attendant console group
- A Terminating Extension Group
- An extension for a hunt group, split, or skill
- An EAS agent’s Login ID
- A VDN Extension

Automatic callback for busy trunks

You can administer your system to call users back if they try to place an outgoing call over a trunk group where all trunks are busy. This is sometimes called Ringback Queueing.

If a multiappearance telephone user has an idle Automatic Callback button and tries to access an all-trunks-busy trunk group, the call is queued automatically. The lamp associated with the Automatic Callback button lights and confirmation tone is heard.

Ringback Queuing is automatic for a single-line telephone. After dialing is complete, the user hears confirmation tone if the queue is available. No action is required.

The system will queue as many calls as allowed based on the Queue Length field on the each trunk group screen. The system checks the busy/idle status of the trunk group just once. If all trunks are busy, the call queues, even if a trunk has become available by the time the caller has completed dialing. This occasionally results in the caller being called back immediately after receiving confirmation tone and going on-hook.

Considerations

- The system cancels an automatic callback request for any of the following reasons:
  - The called party is not available within 30 minutes.
  - The calling party does not answer the callback call within the administered interval (2–9 ringing cycles set in the Automatic Callback-No Answer Timeout Interval field on the Feature-Related System Parameters screen).
  - The calling party decides not to wait and presses the same automatic callback button a second time (multi-appearance telephone) or dials the automatic callback cancellation code (single-line telephone).
- automatic callback is administered to individual telephones by their COS and cannot be assigned to the attendant(s). Multi-appearance telephones must have an automatic callback button to activate the feature.
- automatic callback works differently depending on if the called party was busy or did not answer the call. For a busy call, automatic callback takes place as soon as the called party hangs up. If the called party did not answer, the telephone must be used for another call and then hung up before automatic callback is activated.

  NOTE:
  If the automatic callback originator has all line appearances occupied when the automatic callback call comes in, the user will hear priority ringing once, and the automatic callback lamp will blink. However, if the user presses the automatic callback button to answer the automatic callback call, one of the other calls will drop.

Interactions

- Attendant Call Waiting and Call Waiting Termination

  If a user activates automatic callback to or from a single-line telephone, Call Waiting Termination is denied.

- Attendant Intrusion

  Attendant Intrusion does not work if a user has activated automatic callback.

- Bridged Call Appearance

  Users cannot activate automatic callback from a bridged call appearance. If a user activates automatic callback from a primary extension number, the return-call notification rings at all bridged call appearances.

- Busy Verification

  If a telephone has activated automatic callback, you cannot perform Busy Verification of that telephone.

- Call Coverage

  Automatic callback calls do not redirect to coverage.

- Call Forwarding

  If the called telephone has Call Forwarding activated, the calling party cannot activate automatic callback. However, if automatic callback was activated before the called telephone user activated Call Forwarding, the system redirects the callback call attempt toward the forwarded-to party.

- Call Pickup

  A group member cannot answer a callback call for another group member.

- Class of Restriction

  Telephones with origination restriction cannot activate automatic callback.

- Conference and Transfer

  A single-line telephone user cannot activate conference or transfer if automatic callback is active.

- DCS

  Automatic callback operates over a DCS network as if it were on a local switch.

- Expert Agent Selection

  Users can’t activate Automatic Callback to an EAS agent’s Login ID. They can activate Automatic CallBack to the phone where the agent is logged in.

- Hold

  A single-line telephone cannot receive automatic callback calls if the user has placed a call on hold.

- Hot Line Service

  Telephones administered for Hot Line Service cannot activate automatic callback.

- Intercom - Automatic and Dial

  Intercom calls are not eligible for automatic callback.

- Internal Automatic Answer (IAA)

  IAA does not automatically answer automatic callback calls.

- Manual Originating Line Service

  Telephones with Manual Originating Line Service cannot activate automatic callback.

- Ringback Queuing

  Users can press an automatic callback button to activate Ringback Queuing.

- Telephone Display

  When the system generates an automatic callback call, the display of the originating telephone displays *automatic callback* (or the equivalent for Administrable Language Displays).

Automatic circuit assurance

Automatic circuit assurance (ACA) helps you identify possible trunk malfunctions. With ACA enabled, the system measures the holding time of each trunk call. If the measurements show calls with either extremely long or extremely short holding times, MultiVantage places a referral call to an attendant or telephone.

The system records holding time from when a trunk is accessed to when it is released. You set short-holding-time and long-holding-time limits for each trunk group. The system then compares the recorded holding times against these limits.

You enable ACA for the entire system, and administer thresholds for individual trunk groups. You can have all trunks or only certain trunks measured.

MultiVantage deals with long-holding and short-holding calls differently. For every call that is shorter than the administered short-holding time, the system increases the short-holding counter by 1. For calls over the same trunk that are within the normal range, it decreases the short-holding counter by 1. Thus, trunks that handle a normal variety of call lengths are not singled out as faulty. If the counter reaches the administered short-holding threshold, the system places a referral call.

If one long call exceeds the long-holding time, the system makes a referral call.

You cannot measure personal CO lines, out-of-service trunks, or trunks undergoing maintenance testing.

The referral call

The display or voice-synthesized message that accompanies an ACA call contains the following information:

- The fact that this is an ACA call
- The trunk access code, trunk group number, and trunk group member number
- The type of referral (short or long holding time)

Once the referral call is answered, this information is displayed and remains displayed until the call is released. If the call is not answered within three minutes, the call stops. The system places the call again after one hour, and continues to place the call hourly until someone answers.

The attendant or telephone user who receives the referral call can stop further calls by pressing the aca-halt button, if one is provided. This is a toggle button, and turns off the feature until the user presses the button again.

The audit trail

Each time a referral call is necessary, the system also adds a record to an audit trail. Audit trail records are available on the ACA Measurements Report. Each record contains the following information:

- Time and date of referral
- Trunk group number, trunk access code, and trunk group member
- The type of referral (short or long holding time)

Interactions

- Administrable Language Displays

  You cannot administer languages for ACA messages.

- AUDIX

  Do not set the referral-call extension to a telephone that covers to AUDIX. AUDIX could potentially overload with the volume of calls, because ACA calls remain active for up to three minutes.

- Busy Verification

  Once you have identified a potentially defective trunk, you can use Busy Verification to check it.

- Centralized Attendant Services (CAS)

  When CAS is activated, the referral-call destination must be on the local switch. The system interprets a referral destination of 0 as the local attendant, if one exists. The CAS attendant cannot activate or deactivate ACA referral calls at a branch location.

- Distributed Communications System (DCS)

  Referral calls may be placed across a DCS network. One switch (the primary) is administered to receive ACA referred calls from remote nodes for all switches within the network. You must administer the ACA Remote PBX Identification field on the Feature-Related System Parameters screen with the PBX ID of the node that is designated as primary.

  If ACA referral calls are sent off the switch that generates the referral, the display and voicing information indicating the failed trunk is lost, even if the referral call is made over a DCS network.

- Night Service

  Referral calls to the attendant are not placed if the system is in Night Service mode.

- Visually Impaired Attendant Service

  If the attendant presses the Display Status button and an ACA call has not been answered, then the words Automatic Circuit Assurance are voiced.

  If a visually-impaired attendant presses the Display Status button and the ACA call has been answered, then the words Automatic Circuit Assurance and the extension assigned to the ACA call are voiced.

If your switch contains a voice-synthesis board, ACA referral calls are accompanied by an audible message identifying the type of ACA infraction encountered. The message is “Automatic circuit assurance <long> or <short> holding time threshold has been exceeded for trunk group <#> member number <#>.”

- Voice Message Retrieval

  If you use Voice Message Retrieval, you can assign a nondisplay telephone as a referral destination.

- Wideband Switching

  ACA treats wideband-trunk calls as a single-trunk call and therefore triggers a single referral call. The call information shows the lowest B-channel trunk member associated with the wideband channel.

Automatic customer telephone rearrangement

Automatic Customer Telephone Rearrangement (ACTR) allows a phone to be unplugged from one location and moved to a new location without additional switch administration. The switch automatically associates the extension to the new port. ACTR works with 6400 Serialized phones. The 6400 Serialized phone is stamped with the word “Serialized” on the faceplate for easy identification. The 6400 Serialized phone memory electronically stores its own part ID (comcode) and serial number. ACTR uses the stored information and associates the phone with new port when the phone is moved.

ACTR is an enhancement to Terminal Translation Initialization (TTI), Personal Station Access (PSA), Customer Telephone Activation (CTA). ACTR makes it easy to identify and move phones.

CAUTION:

When a phone is unplugged and moved to another physical location, the Emergency Location Extension field must be changed for that extension or the USA Automatic Location Identification data base must be manually updated. If the Emergency Location Extension field is not changed or if the USA Automatic Location Identification data base is not updated, the DID number sent to the Public Safety Network could send emergency response personnel to the wrong location.

Detailed description

On the Feature-Related System Parameters screen, set the Terminal Translation Initialization (TTI) Enabled field to **y** and the TTI State field to **voice**.

NOTE:
When a phone is moved, if there is any local auxiliary power (a power supply plugged into a local AC outlet), the phone must be plugged into an AC outlet at the phone's new location. A phone with remote auxiliary power must be supplied remote auxiliary power at its new location. If you do not supply auxiliary power in either case after a phone is moved, some optional adjuncts (for example, an expansion module) do not operate.

When you enter **always** or **once** in the Automatic Moves field on the station screen, the switch adds the extension to its ACTR Move List database. When the phone is plugged in, the switch asks the phone for its serial number and records the serial number on the ACTR Move List. If you change the entry in the Automatic Moves field from **always** or **once** to **no**, the switch removes the extension from the Move List.

Call processing

When a phone is unplugged while on a call, and a 6400 Serialized phone that is administered for automatic moves is plugged into the port within 60 seconds:

- both extensions are placed in idle state
- active calls on either extension are dropped, unless the call is active on a bridged appearance at some other phone
- held calls remain in a hold state
- any calls ringing on either extension instantly proceed to the next point in coverage or station hunting path, unless the call is ringing on a bridged appearance at some other phone
- user actions that were pending when the new phone was plugged in are aborted

Design considerations

You can use the **list station movable** command to keep track of extensions on the move list. Once you reach the maximum number, the switch does not allow additional extensions.

Considerations

ACTR can help you troubleshoot phone problems. For example, you can determine if problems originate with the 6400 Serialized phone.

You can unplug a movable 6400 Serialized phone and plug in a maintenance replacement that does not have move permission. If you are able to place and receive calls from the replacement phone, the problem could be with the original phone.

To troubleshoot phone problems with a 6400 Serialized phone using ACTR, one movable phone and one unassigned phone of the same model:

1. Type **change station *nnnn***, where ***nnnn*** is the extension of the 6400 Serialized phone that is not working and press RETURN.

   The Station screen appears.

2. Set the Automatic Moves field to **always** or **no** and press ENTER.
3. Unplug the phone.
4. In the same jack, plug in a phone that does not have move permission of the same model and series into the same jack.

   The switch treats the new phone as a maintenance replacement and does not change the extension and move list administration.

5. Try to place and receive calls from the new phone.

The first phone is the problem if the replacement phone works properly.

The circuit pack is the problem if the replacement phone works incorrectly and experiences the same problems as the first phone.

If you do not have an unassigned phone:

- Change the Automatic Moves field on the Station screen to **no** for the suspect phone and plug it into a different jack.
- Change the Automatic Moves field on the Station screen to **no** for a different phone, then plug it into the suspect jack.

Remember to change the field back when you are done, and reassociate the phone with Customer Telephone Activation (CTA), TTI, or at the switch. If you try either test without changing the field, the extensions move to the new port.

Interactions

Attendant Console

If the attendant console is unplugged and another DCP phone with move permissions is plugged into the port, the new DCP phone’s extension do not move to the port. The port keeps the attendant console’s extension.

Backup Console

Queue threshold warning tone signals every 10 seconds to certain stations. If the set is ACTR moved while the extension is receiving warning tone, or if it becomes eligible to receive the tone by the time it is plugged in, the tone sounds within 10 seconds of the phone being plugged into new port.

Distributed Communications Systems, Uniform Dial Plan

ACTR is not transparent across switches in Distributed Communications Systems (DCS) or Uniform Dial Plan (UDP). The extension number is not carried along when a phone is moved from one switch and plugged into a different switch.

Emergency Access to the Attendant

If the emergency access redirection extension phone is unplugged and another DCP phone with move permissions is plugged into the port, the new DCP phone does not have dial tone. The DCP phone’s extension does not move to the port. The port keeps the emergency access redirection extension.

Expert Agent Selection (EAS)

If an ACTR move takes place, a logged-in EAS agent is automatically logged off.

Night Service

If the night service phone is unplugged and another DCP phone with move permissions is plugged into the port, the new DCP phone does not have dial tone. The DCP phone’s extension does not move to the port.The port keeps night service extension.

Outgoing trunk Queuing (automatic callback, ringback queuing)

If a station ACTR moves before the callback occurs, the callback rings at the new port. If the extension is in x-port (still being moved), the outgoing call attempt is removed from the queue.

Person Station Access (PSA)

If you use PSA to disassociate an extension that is on the move list, the extension stays on the move list, but the serial number for that set is removed the port is X-port.

If you use PSA to associate an extension with a new set, and the extension is on the move list, the serial number on the move list changes to the serial number for the new set.

If you use PSA to associate an extension with a non-ACTR phone the extension is removed from the move list.

Survivable Remote EPN (SREPN)

An ACTR move made under PPN control is recorded in switch translations, but not in the SREPN and vice versa. Do not use ACTR when the switch is running under control of SREPN. SREPN records need to be updated to reflect any ACTR moves made on the PPN.

Terminal Self Administration (TSA)

If a phone in TSA mode is unplugged and another DCP phone with move permissions is plugged into the port, the new DCP phone is idle, not in TSA mode.

Terminal Translation Initialization (TTI)

You can administer an extension, port, or X-port, and add them to the move list, without a port assignment, or a physical phone connected to the port. If a non-serialized 6400 phone is later plugged into the port or associated with that extension, the extension is removed from the move list and the “Automatic Moves” field is set to no. If a 6400 Serialized phone is later associated with that extension that phone’s serial number and port are added to the move list for that extension.

Wan Spare Processor (WSP)

An ACTR move made under PPN control is recorded in switch translations, but not in the WSP, and vice versa. Do not use ACTR when the switch is running under control of WSP. WSP records need to be updated to reflect any ACTR moves made on the PPN.

Automatic Number Identification

Inband Automatic Number Identification

Your switch uses Inband Automatic Number Identification (ANI) to interpret calling party information, such as a calling party number or a billing number.

Inband signaling is when information such as the address digits for the called party is delivered over the same trunk circuit used for the voice or data connection. (Out-of-band or ISDN signaling is when signaling information passes through a different signaling path than the one used for the voice or data connection.)

When a call is made from 555-3800 to your display phone at 81120, and the Incoming Tone (DTMF) ANI field is set to ***ANI*DNIS***, your trunk group receives *5553800*81120*. If the field is set to **ANI*DNIS***, your trunk group receives 5553800*81120*. In both cases, *Call From 555-3800* appears on your display.

If you do not use Inband ANI, the incoming trunk group name appears on your phone display.

Related topics

Refer to ‘‘Trunk Group’’ on page 1167 for information about setting Inband ANI on the trunk group screen.

Outgoing Automatic Number Identification

Outgoing automatic number identification (ANI) applies to outgoing Russian multi-frequency (MF) ANI, R2-MFC ANI and Spain Multi Frequency Espana (MFE) ANI trunks only.

Use Outgoing ANI to specify the type of ANI to send on outgoing calls. You can define MF ANI (the calling party number, sent via multifrequency signaling) prefixes by class of restriction. This allows a switch to send different ANIs to different central offices.

For a tandem call that uses different types of incoming and outgoing trunks, the switch uses:

- the incoming trunk’s COR -assigned call type for Russian or R2-MFC outgoing trunks
- ARS call types for MFE outgoing trunks

Interactions

- Attendant Console

  If an attendant extends a call, the attendant's COR is used to select ANI.

- Authorization Codes

  The authorization code COR is not used to select the ANI. The extension's ANI is used if an extension originates the call, and the ANI for the switch is used if the originating endpoint is an incoming trunk.

- Bridged Call Appearance

  A call from a bridged call appearance uses the ANI of the primary extension.

- Call Vectoring

  The ANI of the originating party is used, not the ANI of the call vector, when a call vectoring route-to command routes a call over an outgoing trunk.

- Distributed Communications System (DCS)

  In a DCS, the ANI sent to the CO is determined by the ANI for PBX on PBX_B, but the category sent to the CO is determined by the Category for MF ANI field on the Class of Restriction screen for the incoming DCS trunk or by the type of call.

- Expert Agent Selection (EAS)

  The EAS agent's login extension and COR is used to determine ANI.

- Hunt Groups and Automatic Call Distribution (ACD) Splits

  The phone's extension and COR is used to determine ANI for a hunt group or ACD split.

- Multimedia Call Handling (MMCH)

  For call origination, multimedia complexes use the COR assigned to their phones.

- Personal Station Access (PSA)

  For ANI, the PSA extension and COR overrides the phone's extension and COR.

- Remote Access

  A remote access barrier code COR is not used for ANI. The extension's ANI is used if an extension originates the call, and the ANI for the switch is used if the originating endpoint is an incoming trunk.

Related topics

Refer to “Displaying ANI calling party information” on page 179 for instructions on setting up ANI display in the U.S.

Refer to “Multifrequency Signaling” on page 1652 for more information on signaling.

Refer to “CLAN and processor CLAN QoS and CIDR support” on page 1521 for more information on Class of Restriction.

Automatic routing — general

MultiVantage automatically routes outgoing calls using the most preferred (normally the least expensive) route available at the time the call is placed. Generally, Automatic Alternate Routing (AAR) routes calls over a private network and Automatic Route Selection (ARS) routes calls using the public network numbering plan. However, both AAR and ARS support public and private networks.

Figure 57 shows you an overview of automatic routing.

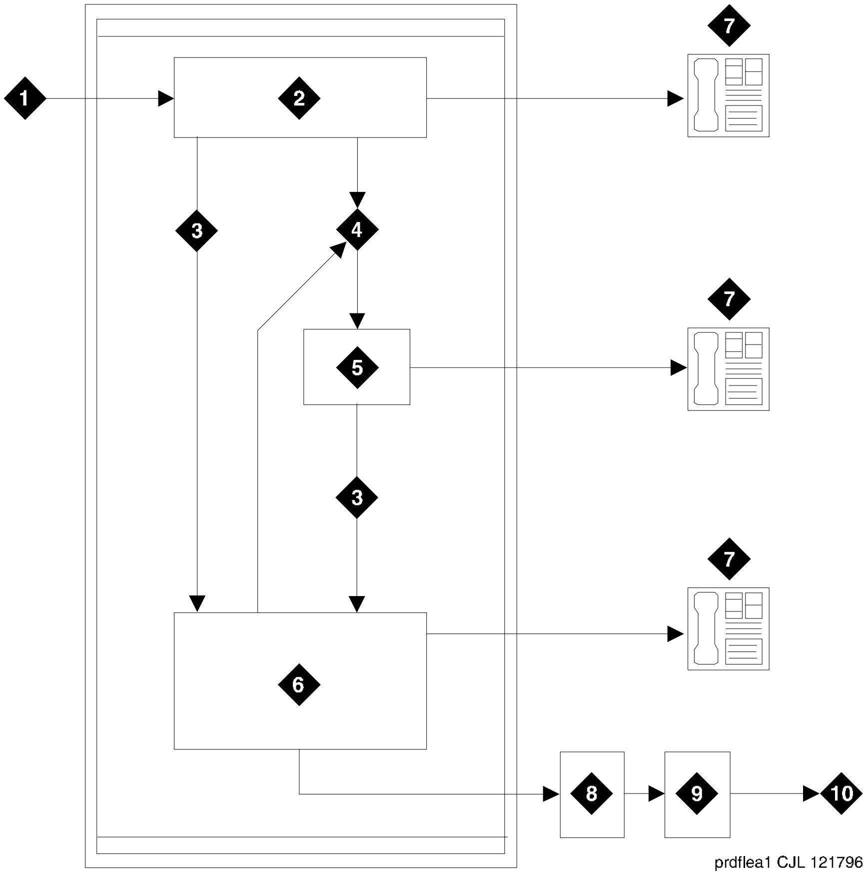


Figure Notes

1. Input from phone, public network trunk, or private network trunk
2. Analyze digits to determine address type (Dial Plan Analysis Table)
3. Direct to AAR/ARS
4. Direct to Uniform Dial Plan (UDP)
5. Analyze digits using UDP to determine route
6. Delete and insert digits (AAR and ARS Digit Conversion Tables)
7. Terminate call at phone
8. Analyze digits (AAR and ARS Digit Analysis Tables) and determine route pattern (Route Pattern, Node Number Routing, Extended Trunk Access screens)
9. Select outgoing trunk group and delete and insert digits
10. Output to public network trunk or private network trunk

Figure 57. Automatic Routing

AAR

AAR routes calls over private networks. When a user dials the AAR feature access code (normally 8 in North America) and phone number, AAR selects the least expensive route for the call in the private network and performs any digit conversion. If the first-choice route is not available, another route is chosen automatically.

AAR routes private-network numbers, public-network numbers, service codes, an international number, operator access code, or an operator-assisted dialing number. AAR routes calls route as far as possible over the private network, and then accesses the public network. This saves long-distance charges and allows you to use your private network as much as possible.

ARS

ARS routes calls over the public network. When a user dials the ARS feature access code (normally 9 in the US and 0 outside of the US) and phone number, ARS selects the least expensive route for the call when there are one or more long-distance carriers or services.

ARS, like AAR, routes private-network numbers, public-network numbers, service codes, an international number, operator access code, or an operator-assisted dialing number, and also routes to Inter-exchange carriers (IXC). These are your long-distance providers.

You can route ARS calls to a variety of types of public-network and private-network trunk groups including Central Office (CO), Foreign Exchange (FX), Integrated Services Digital Network (ISDN), Tie, and Wide Area Telecommunications Service (WATS). See ‘‘Managing trunks’’ on page 403 for more information.

AAR and ARS digit analysis default translations

Your switch contains built-in AAR and ARS Digit Analysis Default Translations. These default translations are used for call processing whether or not AAR or ARS is enabled on your switch.

Any 7-digit dialed string that begins with any number 2 through 9 is processed as an AAR call. Exceptions are listed in the ARS Digit Analysis Default Translations Table. The translations shown on the ARS Digit Analysis Default Translations table are displayed in sorted order (including additions) on each of the 8 possible ARS Digit Analysis Tables.

NOTE:
For service outside of North America, these defaults should be deleted. You can delete the defaults by entering **change ARS analysis 0**. Then use spaces to blank out all of the Dialed String entries.

Table 24. ARS Digit Analysis Default Translations

Dialed String	Total Digits Min.	Total Digits Max.	Route Pattern	Call Type
0	1	1	deny	op
0	8	8	deny	op
0	11	11	deny	op
00	2	2	deny	op
01	9	17	deny	iop
011	10	18	deny	intl
1010XXX0	8	8	deny	op
1010XXX0	18	18	deny	op
1010XXX01	16	24	deny	iop
1010XXX011	17	25	deny	intl
1XXX555	11	11	deny	fnpa
1XXX976	11	11	deny	fnpa
18000555	11	11	deny	fnpa
1809	11	11	deny	fnpa
1900555	11	11	deny	fnpa
411	3	3	deny	svc
555	7	7	deny	hnpa
611	3	3	1	svc
811	3	3	1	svc
911	3	3	1	svc
976	7	7	deny	hnpa
N	7	7	2	hnpa

Continued on next page

Table 24. ARS Digit Analysis Default Translations (Continued)

Dialed String	Total Digits Min.	Total Digits Max.	Route Pattern	Call Type
1N00	11	11	deny	fnpa
1NX	11	11	deny	fnpa

Legend:

N - 2 through 9
X - any digit (0 - 9)
deny - deny
fnpa - foreign number plan area (10-digit call)
hnpa - home number plan area (7-digit call)
intl - international
iop - international operator
op - operator
svc - service

Trunking facilities

MultiVantage can serve as an electronic tandem network (ETN). An ETN is a network of privately-owned trunk and switching facilities that provide a cost-effective alternative to long-distance calling between locations. Each switching facility in an ETN has a unique private-network office code consisting of 1–8 digits.

Traveling Class Marks (TCM) represent the caller’s Facility Restriction Level (FRL) or the FRL of the caller’s access trunk group, and are sent with AAR and ARS numbers sent on ETNs.

Routing with ISDN and overlap sending

You can turn on overlap sending to work on AAR and ARS calls that are routed over ISDN trunk groups. Overlap sending sends ISDN call-address information one digit at a time instead of all the address information going out in one block. This significantly decreases call setup time in countries with complex public-network numbering plans, and is most useful for tandemed calls. See ISDN-PRI Trunk Group and ISDN-BRI Trunk group screens for information on how to set up overlap sending.

Interactions

- Abbreviated Dialing

  The switch does not check the FRL on an AAR or ARS call that uses a privileged Abbreviated Dialing Group List.

Related topics

- AAR and ARS Digit Analysis Table
- AAR and ARS Digit Conversion Table
- AAR and ARS partitioning
- ARS Toll Table (ARS only)
- Class of Restriction
- Dial Plan Analysis Table
- Feature Access Codes
- Generalized Route Selection
- Look-Ahead Routing
- Node Number Routing
- Route Pattern
- Remote Home Number Plan Area
- Time of Day Routing Plan
- Toll Analysis (ARS only)

Barrier codes

A barrier code is a security code used with Remote Access to prevent unauthorized access to your system.

Brief description

Each barrier code must have a Class of Restriction (COR) and Class of Service (COS) assigned. Remote Access has inherent risks; it can lead to large-scale unauthorized long-distance use. To increase your system's security use a 7-digit barrier code with Remote Access Barrier Code Aging. You can administer Remote Access Barrier Code Aging to:

- Limit the length of time an access code remains valid and/or
- Limit the number of times an access code can be used

A barrier code automatically expires if an expiration date or number of accesses has exceeded the limits you set. If both a time interval and access limits are administered for a barrier code, the barrier code expires when one of the conditions is satisfied.

You determine the barrier code length, the actual barrier code, and the barrier code expiration date on the Remote Access screen. You must administer expiration dates and access limits for each of the possible 10 barrier codes. If your system has more than 10 Remote Access users, they must share codes.

When you no longer need a barrier code, remove it from the system. Barrier codes should be safeguarded both by you and their users. If you use barrier codes for outside calls, change them often.

If barrier codes are administered, a special answer-back tone causes a calling modem to leave dial mode. A modem's dialer is sometimes used to gain access (this tone also cancels echo suppressors in the network, preventing DTMF tones from breaking dial tone from a switch). Barrier codes can be used alone or with authorization codes.

Use the **status remote-access** command to view the status of a Remote Access barrier code.

NOTE:
Barrier codes are *not* tracked by Call Detail Recording (CDR). Barrier codes are incoming access codes, whereas, authorization codes are primarily outgoing access codes.

Related topics

Refer to ''Remote Access'' on page 1680 for more information.

Bridged Call Appearance

Bridged Call Appearance allows single-line and multiappearance telephone users to have an appearance of another user's primary extension number. The bridged call appearance can be used to originate, answer, and bridge onto calls to or from the other user's primary extension number.

An appearance of a telephone's primary extension number at another telephone is called a bridged call appearance. A bridged call appearance can be used to originate, answer, or bridge onto an existing call to or from the primary telephone user's extension number. A virtual extension cannot be a bridged call appearance.

On single-line telephones, Bridged Call Appearance is used by going off-hook. On multiappearance telephones, Bridged Call Appearance is used by going off-hook and pressing the bridged appearance button. In both cases, the user is then bridged onto the primary telephone's extension number and can handle calls on that extension number.

An incoming call rings the primary extension number's telephone and all telephones that have a bridged call appearance of the telephone's primary extension number. Each telephone is visually alerted for all bridged appearances on the telephone, but has the option of audible ringing.

On multiappearance telephones, a bridged call appearance can be assigned to any 2-lamp button. It does not require the use of a regular call appearance.

A bridged call appearance can be used just like a regular call appearance for most features. For example, Conference, Transfer, Hold, Drop, and Priority Calling can be used from a bridged appearance, just as they are used from a regular call appearance.

You can administer a telephone with zero call appearances of its primary extension. In this way, a telephone can be administered to have only bridged appearances.

Extension administrable buttons and lamps for multiappearance telephones

You can administer the message lamp and some feature buttons to apply to a specified extension rather than the extension of the telephone they reside on.

- You can administer the message lamp to light when messages are waiting for the extension specified on the Station screen. In this way, the bridged user's telephone can be set up to indicate when messages are waiting for the primary extension.
- You can administer the call forwarding all calls and call forwarding busy/don't answer buttons to activate Call Forwarding for any extension that is on the telephone, even if this extension is a bridged appearance. In addition, you can administer the lamp associated with the call forwarding button to track the call forwarding status of any extension. In this way, a bridged user can activate or deactivate Call Forwarding for all primary and bridged appearances of the extension from the bridged appearance telephone, and the bridged appearance telephone shows the call forwarding status of the specified extension.
- You can administer the send all calls button to activate Send All Calls for any administered extension. The lamp associated with Send All Calls tracks the status of the administered extension. In this way, a bridged user can activate Send All Calls for the primary extension user.

Considerations for single-line telephones

- The number of bridged call appearances allowed varies by system. See *Avaya MultiVantage™ Hardware Solutions Guide* for those numbers. A bridging user cannot have more than one bridged appearance for a particular primary telephone. However, a multiappearance bridging user can have appearances of more than one analog telephone on their telephone (a multiappearance bridging user, by use of different buttons, can bridge onto several different primary telephones).
- The number of bridged appearances allowed on a multiappearance bridging user's telephone is limited only by the number of 2-lamp buttons available on the telephone.
- If the primary single-line telephone is correctly administered, but not in service, calls can still be placed by the bridging users, and received on the bridged appearances of the telephone. The primary telephone can be out of service for several reasons, such as an unplugged telephone, a nonexistent telephone system technician busyout command, etc.
- If more than one user goes off-hook on a bridged appearance at the same time, only the user who was the first to go off-hook can dial.
- If a bridging user is not active on a call, and bridges onto the appearance of an active call, the user is bridged onto the active call. If a multiappearance bridging user is active on a call, and bridges onto the appearance of an active call, the previously selected call is dropped and the user is bridged onto the active call.
- The Privacy-Manual Exclusion feature can be activated by the bridging user only, while active on a call, to prevent accidental bridging of an active call.
- If a call terminates at a telephone on an extension number other than the primary extension number (for example, terminating extension group (TEG), uniform call distribution (UCD) group, call coverage answer group, or direct department calling (DDC) group extension number), a bridged call appearance is not maintained. Therefore, the primary telephone should not be made a member of such a group (even though administration of this is not prohibited).
- The Bridged Call Appearance feature should not be considered as a replacement for Call Coverage or any other similar features.
- If two parties are bridged together on an active call with a third party, and if the conference tone feature is enabled, conference tone is heard.

Considerations for multiappearance telephones

- The number of bridged call appearances allowed at each telephone is limited only by the number of 2-lamp buttons available on the telephone. The number of appearances per primary extension varies by system. See *Avaya MultiVantage™ Hardware Solutions Guide* for this information.
- Up to six parties can be off-hook and involved in a conversation on a bridged appearance of an extension.
- A bridging telephone should have a bridged call appearance corresponding to each call appearance of the primary extension number at the bridged telephone. For example, if a primary telephone has three call appearances, a bridging telephone should have three bridged call appearances of that primary extension. This allows users to refer to the individual call appearances when talking about a specific call.
- Bridged call appearances may result in the reduction of available feature buttons, thereby reducing a user’s capabilities. A Call Coverage module or expansion module can be used to provide up to 20 bridged call appearances. This leaves the other 2-lamp buttons as call appearances, or with other features such as Centralized Attendant Service (CAS).
- If a call terminates at a telephone on an extension number other than the primary extension number (for example, TEG, UCD group, call coverage answer group, or DDC group extension number), a bridged call appearance is not maintained. Therefore, the primary telephone should not be made a member of such a group (even though administration of this is not prohibited).
- Bridged Call Appearance should not be considered a replacement for Call Coverage.
- You can administer conference tone, which, when enabled, is heard when two parties are bridged together on an active call with a third party.

Interactions

- Abbreviated Dialing

  A user, accessing Abbreviated Dialing while on a bridged call appearance, accesses their own Abbreviated Dialing lists. The user does not access the Abbreviated Dialing lists of the primary extension associated with the bridged call appearance.

  A user cannot use an abbreviated dialing FAC after using a priority calling FAC.

- Adjunct Switch Applications Interface (ASAI)

  If you are using ASAI, do not administer more than 16 bridged appearances.

- Attendant Display and telephone Display

  A call from the primary extension number or from a bridged call appearance of the primary extension number is displayed as a call from the primary extension number (the call is displayed as coming from the primary extension number regardless of which appearance placed the call).

  On multiappearance telephones, the display at a principal shows the same information for a bridged call appearance as it does for a nonbridged call. For calls to the principal's extension number, the display at a zero call appearance bridging telephone shows a call from the originator to the principal with no "redirection reason" character. As telephones bridge onto the call, the display updates to show the number of parties in the conference.

- Automatic Call Distribution

  Bridged appearances cannot be accessed via non-ACD hunt groups (although administrable).

- Automatic Callback

  Automatic Callback calls cannot originate from a bridged call appearance. However, when Automatic Callback is activated from the principal user's telephone, the callback call rings (with priority call distinctive ringing signal) at all bridged appearances of the extension as well as at the principal user's telephone. Displays at all telephones (principal and bridged users) show that it is a callback call.

- Busy Indicator (multiappearance telephones only)

  The call is placed to the resource from the first available bridged call appearance for zero primary call appearance telephones.

- Call Coverage

  — Single-line telephones

    When an analog telephone is administered as a bridged call appearance, the telephone user cannot invoke Send All Calls for the extension of their telephone. The user does not have a send all calls button, and the call appearance is associated with another extension. When the user dials a feature-access code (FAC), Send All Calls is activated for the extension associated with the call appearance.

  — Multiappearance telephones

Coverage criteria for bridged call appearances is based entirely on the criteria of the primary extension associated with bridged appearance. A call to the primary extension that requires call coverage treatment follows the coverage path of the primary extension and not the path of any of the bridged appearances. Bridged call appearances do not receive redirection notification.

A user with bridged call appearances can activate or deactivate Send All Calls for a principal's primary call appearance if they are on the bridged appearance.

The primary telephone should not be a member of a call coverage group, because calls to the primary telephone as a member of the group are not bridged.

You can administer the system so that a call can appear at a telephone as both a bridged call and a redirected call. In this way, if the bridged user is the first coverage point, the call redirects to that telephone when the coverage criteria are met.

If the principal is an analog telephone with a bridged call appearance on a multiappearance telephone, an incoming call to the analog telephone that goes to coverage terminates at a primary call appearance on the bridging user's telephone as a coverage call. If the bridging user is a zero primary call appearance telephone, the call cannot redirect to the bridging user since there are no primary call appearances. Therefore, the call redirects to the next available coverage point.

- Call Detail Recording

  If a bridging user originates or answers a call on a bridged appearance, the extension of the bridge is recorded as the calling/called telephone. A conference or transfer by a bridging user also appears as though it was performed by the telephone user.

  On multiappearance telephones, when a call originated from a bridged call appearance on a telephone administered for zero primary call appearance is recorded by Call Detail Recording (CDR), the extension number associated with the appearance is recorded as the calling party. A conference or transfer by a bridged call appearance on a zero primary call appearances telephone also appears as though it were performed by the extension associated with the appearance.

- Call Forwarding All Calls, Call Forward Busy/Don’t Answer

  Call Forwarding can be activated or canceled for the primary extension number from any bridged call appearance of that number. When activated, calls to the primary extension number do not terminate at the bridged call appearances, but go to the designated forwarding destination. Bridged call appearances do not receive redirection notification of the call to the primary extension when it is forwarded unless Ringing — Abbreviated and Delayed is administered.

- Call Park

  When a call is parked from a bridged call appearance, it is parked on the primary extension number.

- Call Pickup

  — Single-line telephones

    Calls to the primary telephone, alerting at bridged appearances of the primary telephone, can be picked up by member’s of the bridging user’s call pickup group. This causes all bridged appearances of the call to be dropped.

    Calls ringing at a primary telephone can be picked up by members of the primary telephone’s call pickup group. However, if the primary telephone and the bridging user’s telephone are not in the same call pickup group, the bridging user cannot pick up calls to other members of the primary telephone’s call pickup group.

    Originating on a bridged appearance and dialing the call pickup FAC is interpreted as an attempt to pick up a call from the primary telephone’s call pickup group.

    A bridging user can use Call Pickup to pick up a call that is alerting at a bridged appearance, instead of selecting the bridged appearance button. This causes the call at the primary telephone and all bridged appearances of the call to be dropped.

    If the bridging user has appearances of numerous single-line (primary) telephones (for example, sales, service, and warehouse), and it is not desired that the calls be answered by anyone other than the primary telephone user or the bridging users, the bridging user(s) should not be assigned to a pick up group.

— Multiappearance telephones

If a telephone receives ringing on a bridged call appearance, the incoming call can be picked up by members of that telephone's call pickup group. This causes all bridged call appearances to be dropped. Calls ringing at a primary telephone can be picked up by members of the telephone's call pickup group. However, if the primary telephone and the bridging user's primary telephone are not in the same call pickup group, the bridging user cannot pick up calls to other members of the primary telephone's call pickup group.

Originating on a bridged appearance and dialing the call pickup Feature Access Code (FAC) is interpreted as an attempt to pick up a call from the primary telephone's call pickup group.

A bridging user can use Call Pickup to pick up a call that is alerting at a bridged appearance, instead of selecting the bridged appearance button. This causes the call to terminate on the bridging user's primary extension button, and the primary telephone and all bridged appearances of the call are dropped.

If the bridging user has appearances of numerous telephones (for example, sales, service, and warehouse), and it is not desired that the calls be answered by anyone other than the telephone user or the bridging users, the bridging users should not be assigned to a pick up group.

A telephone with zero primary call appearances can be assigned to a call pickup group.

- Call Waiting Termination (single-line telephones only)

  Call Waiting Termination applies only to an active call on the primary telephone that has no one else bridged on. If one or more bridging users are active on a call, call waiting calls are denied whether or not the primary user is also off-hook on the call. A bridging user can bridge onto a call with the primary user if there is also a call waiting.

- Class of Restriction (multiappearance telephone users only)

  The COR assigned to a telephone's primary extension also applies to calls originated from a bridged call appearance.

- Conference — Attendant, Conference
  - — Single-line telephones

    A bridged call cannot be conferenced if more than one user is active on that call. This is because the bridging user has no access to the call after the primary telephone user places the call on soft hold, and the primary telephone user has no access to the bridging user’s call appearance used for conference/transfer attempts.

    If a bridging user is active on a bridged call and the primary analog telephone user attempts a conference, the attempt is ignored. The same is true if an analog bridging user attempts a conference when the primary telephone user and another bridging user is active on a call.

    When the primary telephone user is active on a call, and no other bridging user is active on the call, that call can be placed on hold by the primary telephone user utilizing normal single-line conference procedures. Any attempt by a bridging user to bridge onto the call during a successful conference attempt is denied.

    A bridging user, alone on a bridged call, can conference the call utilizing the normal multiappearance telephone conference procedures. Any attempt by the analog primary telephone user to bridge onto the call during a successful conference attempt is ignored. Any attempt by other bridging users is denied (standard denial response is returned to the bridged appearance).

    If a conference is not allowed because of the preceding limitations, the user can accomplish a transfer by asking an internal nonbridged party in the connection to create the conference, or asking the remaining bridging users and primary user to disconnect so that the conference can be completed. At completion of the conference, the parties that left the call can reenter the call if control of the conference remains with the primary telephone. If control of the conference does not remain with the primary telephone, the bridging user must conference the primary telephone and the bridging user back into the call as required.

    If the bridging user has no other available bridged appearances of the primary extension (other than the one he or she is currently on), the bridging user, after pressing the conference/transfer button, must select a call appearance to be used for the conference, before dialing the number.

— Multiappearance telephones

Call Waiting Termination applies only to an active call on the primary telephone that has no one else bridged on. If one or more bridging users are active on a call, Call Waiting calls are denied whether or not the primary user is also off-hook on the call. A bridging user cannot bridge onto a call with the primary user if there is also a call waiting.

Conferences can be set up on bridged appearances using the usual conference operations. Either a primary extension button or a bridged appearance button can be used to make the calls for adding to the conference.

You can administer the system to automatically select the first idle appearance if there is no idle appearance with an extension matching the extension that is conferencing the call.

When the user presses the conference button (the second time) to connect the parties together, the newly formed conference call appears on the primary or bridged appearance to which the user was connected at the time of that last conference button depression. The other appearance is disassociated from the conference call. Therefore, if the original call is on a bridged appearance, and the conference is formed on an appearance of the bridged user’s own primary extension, the bridged extension becomes disassociated from the conference call and the principal user of that bridged extension can no longer bridge onto the conference.

This disassociation of the conference from the bridged extension can be avoided by setting up the conference in the opposite order. To do this, the user:

1. Presses the hold button to hold the original call on the bridged appearance
2. Selects a call appearance and calls the party to be added
3. Presses the conference or transfer button
4. Selects the held bridged appearance
5. Presses the conference button (again)

When this procedure is used, the conference is formed on the bridged appearance so that the primary user of the bridged extension can still bridge onto the conference call.

If the primary user and the bridged user are both on the call when one user transfers the call, the user performing the transfer becomes the controlling user for the participation of both users on the conference. To disassociate the appearance from the call, the controlling user must be the latter of the two users to hang up from the call. If the controlling user hangs up first, the appearance goes on soft hold when the noncontrolling party hangs up. In this case, one of two things must occur to disassociate the appearance from the call: all other parties on the call hang up, or the controlling user rejoins the call and hangs up again.

The display shows the number of other active parties in a call, including active bridged appearances.

- Consult (multiappearance telephones only)

  Bridged call appearances of the primary extension do not ring on a consult call to the primary extension.

- Coverage Answer Group

  - Single-line telephones

    The primary (analog) telephone is not a member of a call coverage answer group, because calls to the primary telephone as a member of the group *are not bridged*.

    If the primary telephone is made a member of a coverage group, coverage criteria is based entirely on the criteria of the primary telephone. This means that a call to the primary telephone that requires call coverage treatment follows the path of the primary telephone and not the path of any of the telephones with bridged appearances of the primary telephones. In this case, it is desirable to have the bridging user in the coverage path of the primary telephone. Then, when a call to the primary telephone requires coverage treatment, it follows the coverage path to the bridging user's telephone, call appearances of the call are dropped, and the call terminates at the bridging user's telephone as a coverage call.

  - Multiappearance telephones

    Bridged call appearances of a primary extension do not ring when there is a Coverage Answer Group (CAG) call to the primary extension. Bridged call appearances cannot bridge onto the call.

- Data Privacy, Data Restriction

  When Data Privacy is activated or Data Restriction is assigned to a telephone involved in a bridged call and the primary telephone and/or bridging user attempts to bridge onto the call, Data Privacy and Data Restriction are automatically overridden (or deactivated in the case of Data Privacy).

- Emergency calls

  If a user dials an emergency call from a bridged appearance, the Calling Party Number that is sent to the public safety answering point for US E911 location is the bridged extension, not the extension of the physical phone from which the call is made. If the physical phone is located far from the phone to which it is bridged, the emergency response team may not be able to locate the caller. It is preferable to place E911 calls from an actual call appearance, rather than a bridged appearance.

- Hold — Automatic
  - Single-line telephones

    A call cannot be put on hold if more than one user is active on that call.

    The primary telephone user, when no other bridges are active on the call, can put the call on hold, using normal single-line hold procedures. If the primary telephone user successfully soft holds the call, the status lamp at all of the bridged appearances shows the hold indication; and then the call can be put on hard hold by dialing the hard hold FAC. The hard held call is no longer accessible to the bridging users until it is taken off hold by the primary telephone user. After the call is put on hard hold, any new call to the primary telephone is tracked by the bridged appearances.

    A bridging user can place an active call on hold (if the primary telephone or any other bridges are not active on the call) by using normal multiappearance hold procedures. Any attempt to enter the held call returns it to the status of an active call that can then be accessed using bridging procedures.

    If hold is not allowed because of the preceding reasons, the user can just go on-hook and then reenter the call as required, because the call remains accessible as long as the primary telephone or any bridging user is active on it.
  - Multiappearance telephones

    Any user (primary or bridged appearance) can place an active call on hold. If only one user is active on a call and places that call on hold, the indicator lamp at both the principal’s appearance button and the bridged party’s appearance button shows that the call is on hold. If more than one user is bridged onto the active call, and one of the users activates Hold, the activator receives “hold” indication for the call and status lamp of all other bridged users remains active.

- Hotline Service (single-line telephones)

  If a single-line telephone is administered for Hotline Service, bridged appearances of that telephone's extension also places a hot line call automatically when a user goes off-hook on that bridged appearance.

- Hunt Group (DDC or UCD)

  Bridged call appearances cannot be used in conjunction with DDC or UCD hunt groups.

  Although you can assign a bridged extension to a hunt group, such assignment is not recommended because DDC/UCD calls do not terminate at any bridged appearances of that extension on other telephones.

- Intercom — Automatic and Intercom — Dial (multiappearance telephones only)

  Bridged appearances of a primary extension number are not rung for intercom calls. Furthermore, if a telephone has no primary call appearances it can never be rung for an intercom call. Therefore, if a secretary is screening all calls for the principal, and is indicating who is calling via intercom, the principal must have a call appearance on which to receive and send intercom calls.

- Internal Automatic Answer

  Calls terminating to a bridged appearance of an IAA-eligible telephone are not eligible for IAA.

- Last Number Dialed

  Activation of the LND feature causes the last number dialed from the activating telephone to be redialed, regardless of the extension number used (primary or bridged call appearance).

- Leave Word Calling

  A LWC message left by a user on a bridged call appearance leaves a message for the called party to call the primary extension number assigned to the bridged call appearance.

  When a user calls a primary extension, and activates LWC, the message is left for the primary extension, even if the call was answered at a bridged call appearance.

  LWC messages left by the primary user can be canceled by a bridged appearance user (for example, a secretary can cancel a LWC message left by a boss).

- Personal Central Office Line
  - Single-line telephones

    A single-line primary telephone cannot be a member of a Personal Central Office Line (PCOL) group.
  - Multiappearance telephones

    If a user is active on his or her primary extension number on a PCOL call, bridged call appearances of that extension number cannot be used to bridge onto the call. The call can only be bridged onto the call if another telephone is a member of the same PCOL group and has a PCOL button.
- Preference

  Ringing Line Preference selects an alerting bridged appearance; Idle Line Preference does not.
- Priority Calling

  The primary telephone user or the bridging user can make a priority call. If a priority call is made to an idle telephone, the primary telephone and all bridging users are alerted by priority alerting.

  A user cannot use an abbreviated dialing FAC after using a priority calling FAC.
- Privacy-Manual Exclusion

  Exclusion prevents any other user from bridging onto the call. Activation of exclusion by any user (primary or bridged appearance) before placing a call, prevents any other user from bridging onto the call. Activation of exclusion by any user active on a call, while the primary user and/or any other bridging users are active on the call, drops all other users from the call (including the primary user), leaving only the activator and the calling/called party on the call.
- Redirection Notification (multiappearance telephones only)

  Redirection Notification is not provided at telephones with a bridged appearance of a primary extension number unless Ringing — Abbreviated and Delayed is administered to give notification.
- Ringback Queuing

  Ringback Queuing is not provided on calls originated from a bridged call appearance. However, after the principal user of the bridged extension has activated Ringback Queueing, the resulting callback call alerts at bridged appearances as well as at the principal user’s telephone. The call can be answered from the primary user’s telephone or from any bridged appearance.

- Ringer Cutoff (multiappearance telephones only)

  Ringer Cutoff prevents any nonpriority (or nonintercom) incoming call from ringing at that telephone. This is independent of whether the call is to the telephone's primary extension or to any of the bridged appearance's' extensions.

- Ringing — Abbreviated and Delayed

  See Ringing — Abbreviated and Delayed, for other bridged appearance alerting options.

- Service Observing

  The telephone user or bridging user can bridge onto a service observed call at any time. If the telephone is being service observed and an incoming call is answered by the bridging user, the call is not observed unless or until the telephone user bridges onto the call. Conversely, if the bridging user is being service observed and an incoming call is answered by the telephone user, the call is not observed unless or until the bridging user bridges onto the call.

  If the bridging user activates Service Observing using a bridged appearance, Service Observing is activated for the bridging user.

- Terminating Extension Group

  TEG calls to the primary extension do not ring at the associated bridged appearances. TEG calls cannot be answered or bridged onto from a bridged appearance of the TEG member's primary extension. The primary telephone should not be assigned to a TEG.

- Transfer

  — Single-line telephones

    A call cannot be transferred by an analog telephone if more than one user is active on that call.

    The primary telephone user, when no other bridges are active on the call, can transfer the call using normal single-line transfer procedures. Any attempt by a bridging user to bridge onto this call during a successful transfer attempt is denied (a standard denial response is returned to the bridged appearance).

    An analog bridging user, alone on a bridged call, can transfer the call, using normal transfer procedures. Any attempt by the primary telephone user to bridge onto this call during a successful transfer attempt is ignored; and any attempt to bridge on by a bridging user is denied.

If the bridging user has no other available bridged appearances of the primary extension (other than the one he or she is currently on), the bridging user, after pressing the conference/transfer button, must select a call appearance to be used for the transfer, before dialing the number.

- Multiappearance telephones

  If the bridging user has at least one available bridged appearance of the primary extension (other than the one he or she is currently on), the system automatically selects a bridged call appearance for the transfer when the conference/transfer button is pressed.

  You can administer the system to automatically select the first idle appearance if there is no idle appearance with an extension matching the extension that is transferring the call.

  If the primary user and the bridged user are both on the call when one user transfers the call, the user performing the transfer becomes the controlling user for the participation of both users on the conference. The controlling user is immediately dropped from the call. When the noncontrolling user hangs up, the appearance goes on soft hold. In this case, one of two things must occur to disassociate the appearance from the call: all other parties on the call hang up, or the controlling user rejoins the call and hangs up again.

- Videophone 2500 (single-line telephones)

  A user may not use an analog bridge to a Videophone 2500 principal that is on a video call.

- Voice Message Retrieval

  A voice message to the primary extension can be retrieved on a bridged appearance by the bridged appearance user. If a security code is required to retrieve the message, the bridging user must use the security code of the primary telephone.

- Voice Paging

  The use of Voice Paging automatically invokes exclusion. Therefore, interactions for this feature are the same as for Privacy-Manual Exclusion.

Busy Indicator

The Busy Indicator button provides multiappearance telephone users and attendants with a visual indicator of the busy or idle status of one of the following system resources:

- An extension number
- A trunk group
- A terminating extension group
- A hunt group—either direct department calling (DDC) or uniform call distribution (UCD)
- Any loudspeaker paging zone, including all zones

The Busy Indicator button provides the attendant or user with direct access to the extension number, trunk group, or paging zone.

You can assign extension numbers, trunk group access codes, and Loudspeaker Paging access codes to a Busy Indicator button.

The Facility Busy lamp indication for a vector directory number (VDN) does not light when the VDN is being used. The associated button may be used to place a call to a VDN.

Busy Tone Disconnect

In some regions of the world the CO sends a busy tone for the disconnect message. With Busy Tone Disconnect, the switch disconnects analog loop-start Central Office trunks when a busy tone is sent from the CO.

A call that is originated from or terminated to a phone using a BTD enabled trunk has a Call Classifier port connected to the trunk. The Classifier port connects, then the call is answered and stays connected on the trunk until the station hangs up or a BTD signal is received from the CO. If there is only one BTD trunk on a call when the BTD signal is received, the call is dropped. If it is a conference call, only the trunk is dropped and the rest of the parties stay connected.

Interactions

- Answer supervision

  If Answer supervision is enabled, set the Answer supervision timeout field to 0 (zero).

Busy Verification

Busy Verification (Verify button) allows attendants and specified multiappearance telephone users to make test calls to trunks, telephones, and hunt direct department calling (DDC) and uniform call distribution (UCD) groups. Attendants and multiappearance telephone users can distinguish between a telephone that is truly busy and one that only appears busy because of some trouble condition. They can also use this feature to quickly identify faulty trunks.

An attendant or multiappearance telephone user can activate Busy Verify by pressing the Verify button. If they want to verify a telephone or hunt group, they enter an extension number. If they want to verify a trunk, they dial a trunk-access code, followed by the 2- or 3-digit number of the trunk-group member to be verified. If the trunk-group member number is less than 10, the system requires a leading zero (01 or 001 rather than 1).

NOTE:
For DEFINITY Server SI or DEFINITY Server CSI, the member number is a 2-digit number; for DEFINITY Server R, the member number is a 3-digit number.

After an attendant or multiappearance telephone user has activated Busy Verification, the system checks the validity of the extension or trunk-access code and member number. If the number is not a telephone extension, DDC/ UCD group-extension, Automatic Call Distribution (ACD) split number, or trunk access code with a valid member number, the system denies Verify and returns intercept tone.

When you use Verify to check a valid telephone extension (one that is in the dial plan and assigned to a telephone), the system initiates a priority call to that extension. Table 25 on page 1407 describes the process.

Table 25. Verification of a telephone

Telephone Status	System Response	Result
Idle	■ Generates priority ringing at the telephone ■ Processes the call as a normal telephone-originated or attendant-originated call	■ Verification is complete. ■ Anyone can place a call to the telephone.
Active on a call and has an idle call appearance	■ Generates priority ringing at the first idle appearance ■ Processes the call as a normal attendant-originated call	■ Verification is complete. ■ Anyone can place a call to the telephone extension.
Active on a call and has no idle call appearances or has only one line appearance	■ Bridges the attendant onto the call ■ Generates a warning tone to all active parties and repeats the tone every 15 seconds while the attendant remains bridged onto the call	■ Verification is complete. ■ The attendant can determine if the telephone is actually in use.
Out of service	■ Generates reorder tone	■ Verification is denied.

When you use Verify to check a valid ACD split, UCD group, or DDC group, the system initiates a priority call to that group. (Valid in this case means the split or group is translated and at lease one member is logged in.) Table 26 on page 1408 describes the process.

Table 26. Verification of an ACD Split, UCD Group, or DDC Group r

Split or Group Member Status	System Response	Result
Available for an incoming call	■ Generates priority ringing at the member's telephone ■ Processes the call as a normal attendant-originated call	■ Verification is complete. ■ Anyone can place a call to the member's telephone.
All activated Make Busy	■ Generates reorder tone	■ Verification is denied.
Not available for incoming calls	■ The system does not queue the call even if a queue is available. ■ Generates reorder tone	■ Verification is denied.

When you use Verify to check a valid trunk, the system checks the status of that trunk. (Valid in this case means the trunk is translated with members and is not in an out-of-service state.) Table 27 describes the process.

Table 27. Verification of a Trunk

Trunk Status	System Response	Result
The trunk is idle and incoming.	■ The system generates confirmation tone.	■ Verification is complete. ■ Anyone can use the trunk.
The trunk is idle and outgoing.	■ The system generates dial tone.	■ Verification is complete. ■ Anyone can use the trunk.
The trunk is busy with an active call.	■ The system bridges the Verify originator onto the call. ■ The system generates a warning tone to all active parties and repeats the tone every 15 seconds while the Verify originator remains bridged onto the call.	■ Verification is complete. ■ The trunk is in use.
The trunk is out of service.	■ The system generates reorder tone.	■ Verification is denied.

Considerations

- A busy verification cannot be made to an analog extension that is waiting to be answered at another extension. A call must be answered before it can be verified.
- If your country requires a tone other than 440 Hz, use the Intrusion feature rather than Verify to verify telephones.
- The system does not provide bridging when you verify UCD and DDC groups or RLTs.
- You cannot make outgoing test calls on DID trunks.
- You can verify an extension that is administered without hardware (X-ported). In this case, the system generates reorder tone.

Interactions

- Automatic Callback

  Once the called party in an Automatic Callback call hangs up, neither extension can be busy-verified until both the calling and called parties are connected or the callback attempt is canceled (by the activating party or by time-out of the callback interval).
- Call Coverage

  Since the busy-verification call to an extension is originated as a priority call, the call does not go to coverage.
- Call Forwarding

  Busy verification made to an extension with call forwarding activated, does not busy verify the forwarded-to extension. Only the called extension is busy verified.
- Call Waiting Termination

  You cannot verify an extension that called an active telephone and is receiving call-waiting ringback tone unless the extension has an idle call appearance.
- Conference

  The system denies busy verification of any extension involved in a conference call of more than five people. However, the system does allow a busy verification of any extension involved in a conference call of 5 or fewer parties. The system also denies busy verification of a trunk on a 6-party call.

- Data Privacy

  Busy verification is denied if it would cause a bridging attempt on a telephone that has activated Data Privacy.

- Data Restriction

  The system denies Verify if Data Restriction is active on a call, and a busy verification bridging attempt is made on that call.

- Hold

  Busy verification of a multiappearance telephone is denied if all call appearances have calls on hold.

- Individual Attendant Access

  An attendant cannot make a busy verification of another individual attendant console or of the attendant group.

- Loudspeaker Paging Access

  The system denies busy verification if the telephone or trunk to be verified is connected to paging equipment.

- Transfer

  Once the originator of busy verification has bridged onto a call, any attempt to transfer the call is denied until the originator drops from the call.

- telephone Origination Restriction

  A telephone that is origination restricted can be assigned a Busy Verify button. However, the button cannot be used.

- telephone Termination Restriction

  The system denies busy verification of telephones that are termination restricted.

Related topics

- Attendant Console screen

  Feature Button Assignments
  — verify
  — Display Language

- Station screen (multiappearance phones)

  Button/Feature Button Assignments
  — verify
  — Display Language

Call Charge Information

Avaya MultiVantage provides two ways to know the approximate charge for calls made on outgoing trunks:

- Advice of Charge — for ISDN trunks

  Advice of Charge (AOC) collects charge information from the public network for each outgoing call. Charge advice is a number representing the cost of a call; it is recorded as either a charging or currency unit.
- Periodic Pulse Metering — for non-ISDN trunks

  Periodic Pulse Metering (PPM) accumulates pulses transmitted from the public network at periodic intervals during an outgoing trunk call. At the end of the call, the number of pulses collected is the basis for determining charges.

Call-charge information helps you to account for the cost of outgoing calls without waiting for the next bill from your network provider. This is especially important in countries where telephone bills are not itemized. You can also use this information to let employees know the cost of their phone calls, and so encourage them to help manage the company's telecommunications expenses. Note, however, that you cannot necessarily use this information to dispute telephone bills with the network provider.

You need to request either AOC or PPM service from your network provider. In some areas, your choice may be limited. Your Avaya representative can help you determine the type of service you need.

NOTE:
This service is not offered by the public network in some countries, including the US.

Detailed description

The availability of AOC and PPM varies from one country to the next. In some countries, AOC information is received automatically for each call. In others, the system must request AOC information for each call.

In some countries, the public network sends call-charge information only at the end of a call. In others, the public network sends information during a call as well. PPM is available over the following trunk types:

- CO
- DIOD
- FX
- PCOL
- WATS

CDR Output

The ISDN Call Charge or PPM field contains the last cumulative charge received from the network. If Call Splitting or Attendant Call Recording is enabled, and a call has been transferred for the first time, the ISDN Call Charge field contains the cumulative charge most recently received from the network.

For all subsequent transfers, the ISDN Call Charge field contains the difference between the cumulative charge most recently received and the value generated in the previous CDR record for the same call.

A zero appears in the Call Charge field when: no AOC information is received; a value of zero is the last charge information received; or the outgoing trunk group is not administered for AOC or PPM.

Charge Display and CDR

Avaya MultiVantage provides two ways for you to view call-charge information: on a display or as part of the Call Detail Recording (CDR) report.

From a display, you can see the cost of an outgoing call, both while the call is in progress and at the end of the call. If you want end users to control when they view this information, you can assign a display button that they can press to see the current call charges. If you want call charges to display automatically whenever a user places an outgoing call, you can set the Automatic Charge Display field to **y** on the user's Class of Restriction (COR) screen.

You can administer the system so that call charges appear on CDR reports. You can also allow users to view call charges on phone displays. For information on how to set this up, see ''Viewing call charge information'' on page 535.

Considerations

The Primary (or Secondary) Output Format for the CDR report must be one of the following types.

- Customized — You must also include either the ISDN-CC or PPM fields in your record design. If you use both types of call-charge information, you need both fields.
- Expanded (Enhanced 24-word standard ASCII)
- Int-direct
- Int-ISDN (International ISDN expanded)
- Int-process
- Unformatted (Enhanced 24-word standard ASCII)
- If you want call charges to restart at 0 for calls that are forwarded or transferred, administer Outg Trk Call Splitting.

Attendant consoles

Automatic Charge Display Mode does not apply to attendant consoles. The attendant must always press a button to enter display mode.

Performance impact

Call Charge Information can impact system performance in several ways. The information coming in over ISDN trunks takes up bandwidth, and reduces the maximum amount of traffic the ISDN D-channel can handle. This is especially true in countries such as Germany and France, where the network sends charging information updates as often as every 3 to 10 seconds for each active international call.

The number of sets that display charge information and the frequency of updates also affect performance. Normally, the update frequency should match the average rate at which call charge updates are received from the public network.

CAUTION:

Updating displays too frequently can cause unnecessary system performance degradation. If performance slows to an unacceptable rate, you may need to lengthen the amount of time between updates.

Button operations

If you administer a button for charge display, the display-set user can press the disp-chrg button at any time during the call to see the current charges. If your public network sends charge information only at the end of a call, display-set users must have this button and press it just before they hang up. After the call drops, the charge will appear.

Other display functions

If a user invokes the elapsed-timer display, the timer may overwrite part of the charge display. If the user has a local directory (Dir button), and presses this button while a call is in charge display mode, the call-charge information will overwrite the directory any time an update comes in. To avoid this, the user must press Exit or Normal.

Interactions

- Attendant Features

  Attendant consoles cannot have automatic charge display. If you want the attendant to see call charges, you must assign a disp-chrg button to the attendant console. If the attendant moves to transfer an outgoing call, the display returns to normal mode. If the transfer does not complete, or the call remains at the attendant console for whatever reason, the attendant must press the disp-chrg button again to view call charges.

- Automatic Incoming Call Display

  If a user has charges displayed for an existing call and a second call rings on another line appearance, the display returns to normal mode for a short time to show the identity of the caller. The user must press disp-chrg again to view call charges, or if automatic charge display is enabled, must wait for the Charge Update Frequency Timer to expire.

- Bridged Appearance

  If a user makes a call using a bridged appearance, the call charges display on the telephone from which the call is made. If that telephone has Automatic Charge Display as part of its COR, the charges will appear automatically. The actual charge for the call appears on the CDR report as if the call had been made from the principal's extension, not the bridged appearance.

- Call Coverage or Forwarding — Off Net

  Call charges for a call to an extension whose calls are redirected over a public-network trunk are charged to the called extension, not the calling extension. However, if the call is placed from an internal phone that has charge display capability, the caller will see the charges for the redirected call.

- Call Park

  When a user parks a call, the display mode returns to Normal. If a user retrieves a parked, outgoing call from another display telephone, the display on that set shows the current call charges if the user presses a disp-chrg button, or if the user's COR allows Automatic Charge Display. If call splitting is enabled, the display shows the charges accumulated since the user unparked the call.

- Call Transfer

  For Advice of Charge, if a transferred call is routed over a public-network ISDN-PRI trunk group, AOC administration for the outgoing trunk group controls whether AOC information is requested or recorded for the call. If two or more outgoing trunks are connected together via trunk-to-trunk transfer, the Avaya MultiVantage may receive AOC information from the network for each outgoing trunk involved in the call.

- CDR Adjuncts

  Avaya MultiVantage does not tandem AOC information through a private network to other switches. Therefore, the CDR adjunct that records AOC information must receive its input from Avaya MultiVantage directly connected to the public network.

- CDR Call Splitting

  If you use CDR Call Splitting for outgoing trunks, each time a call is transferred, the system generates a separate record. Attendant Call Recording, a form of Call Splitting, generates a CDR record when an attendant drops from a call. Incoming Trunk Call Splitting has no effect on charge information.

  If you rely on Call Splitting or Attendant Call Recording, you should request call charge information during the call. However, for AOC, this increases message activity on the signaling channel and reduces Busy Hour Call Capacity of the MultiVantage System.

  In some countries, or with specific protocols, AOC information during a call is not available. In this case, you can use the Elapsed Time in the CDR records to allocate the charges among the parties on the call.

  You must use CDR Call Splitting if you want the charge display to restart at 0 when a call is transferred.

- Centralized Attendant Services

  In any configuration where a branch system has no direct connection to the public network, the private network does not pass call-charge information to these branches.

- Conference

  If a user adds a third party to a call in charge-display mode, the display returns to normal. Call charges will not appear as long as there are more than two parties on the call.

- Distributed Communications System (DCS)

  In any configuration where a branch system has no direct connection to the public network, the private network does not pass call-charge information to these branches.

- Electronic Tandem Network (ETN)

  In any configuration where a branch system has no direct connection to the public network, the private network does not pass call-charge information to these branches.

- Hold

  If a user places a call on hold, the display returns to normal mode. The user must press disp-chrg again to view call charges, or if automatic charge display is enabled, must wait for the display to refresh.

- Last Number Dialed

  Users can view the dialed number while active on a call by pressing the stored-numb button, then the last-numb button. To view call charges again, the user must then press the disp-chrg button, or (if Automatic Charge Display is part of the user's COR) the Normal button.

- QSIG

  In any configuration where a branch system has no direct connection to the public network, the private network does not pass call charge information to these branches.

- System Resets

  If you perform a warm reset while calls are active with charge display, the charge display will freeze. To resume call charge updates, users must press the Normal button.

Call Coverage

Call Coverage provides automatic redirection of calls to alternate answering positions in a Call Coverage path. Call Coverage allows you to:

- Establish coverage paths with up to 6 alternate answering positions
- Establish redirection criteria that govern when a call redirects
- Redirect calls to a local switch location
- Redirect calls to a remote (off-net) location
- Redirect calls based on time-of-day
- Allow users to change back and forth between two coverage choices (either specific lead coverage paths or time-of-day tables). Refer to ''Extended User Administration of Redirected Calls'' on page 1554 for more information.

Hardware requirement

The Coverage of Calls Redirected Off-Net (CCRON) generally requires call classification hardware. Both the Call Classifier - Detector and Tone Clock/Call Classifier - Detector circuit packs provide tone detection ports including the capability to do call classification. There are 8 ports on each circuit pack.

For countries using the USA tone plan, a Call Classifier - Detector or Tone Clock/Call Classifier - Detector circuit pack is sufficient to provide call classification.

For countries not using the USA tone plan, the Call Classifier - Detector and Tone Clock with Call Classifier - Tone Detector circuit packs must be configured appropriately to provide call classification.

The number of simultaneous monitored calls depends on the:

- total amount of outbound call traffic,
- number of call classification ports available, and
- use of other switch applications that make use of call classification ports.

Coverage of Calls Redirected Off-Net competes with the following switch applications for ports on the Call Classifier - Detector and Tone Clock with Call Classifier - Tone Detector circuit packs:

- Answer Detection
- Call Prompting
- CallVisor ASAI
- Multi-Frequency Compelled (MFC) signaling

Serious degradation of switch performance, including the inability to launch new calls, can result from an insufficient resource of call classifier ports.

Detailed description

When a call meets the redirection criteria of the principal, the call attempts to route to one of up to 6 points in the coverage path. If no coverage points are available, the call may revert to the called principal or group. If any point in the path is available, the call either rings the individual phone or member of a group specified for that point or queues on the group. Once a call is ringing or queued at any point in a coverage path, the call never reverts to the called principal or group, or to the previous point. A call remains at a coverage point for the Coverage Subsequent Redirection interval. At the end of this interval, the call attempts to route to any remaining points in the coverage path. If no other point is available to accept the call, the call remains queued or continues ringing the current coverage point.

Call Classification

Classifying a call means determining the state of the call at its final destination. That means whether the call was answered, received busy, reorder, intercept, special information tone (SIT), or other treatment.

Call classification is accomplished by the ISDN protocol or ports on the:

- Tone Clock with Call Classifier - Tone Detector circuit pack
- Call Classifier - Detector circuit pack

Coverage Path

A Call Coverage path is a list of up to six alternate answering positions (covering users/points) that are accessed, in sequence, when the called party or group is not available to answer the call.

You can assign any of the following entities a coverage path so they are eligible to have calls redirected to coverage:

- ACD split
- Agent LoginID
- PCOL group
- TEG
- Hunt group
- Phone (on-net or off-net)

You establish the coverage paths and set the redirection criteria. If a coverage path is not assigned to a particular facility, calls are not redirected from that facility, unless another feature is assigned. A coverage path can include any of the following:

- Announcement
- Attendant group
- AUDIX
- Coverage answer group
- Hunt group
- Public network number (off-net)
- Vector directory number (VDN)
- Phone (on-net or off-net)

Avaya MultiVantage allows for multiple coverage paths. However, for any particular call only one coverage path is used. The “lead” coverage path is the first coverage path in a chain that is considered when a call redirects to coverage. The chain is defined in the Next Path Name field on the Coverage Path screen.

When a call redirects to coverage, the lead coverage path at that time is checked to determine whether its coverage redirection criteria match the call status. If there is a match the lead coverage path is used. If the lead coverage path’s redirection criteria does not match, the system moves down the path chain until it finds a coverage path with redirection criteria that matches the call status. If the chain is exhausted before the system finds a match, the call does not redirect to coverage. Once a coverage path is selected, it is used exclusively through the duration of the call.

You can assign lead coverage paths directly in the Coverage Path 1 or Coverage Path 2 fields on the appropriate screens. For example, to assign a lead path for a TEG, set the Coverage Path field on the Terminating Extension Group screen. You can also assign the lead paths indirectly by assigning a Time-of-Day Coverage Table to the Coverage Path 1 and Coverage Path 2 fields. Then, the system selects the lead path according to the time of day.

Subsequent redirection interval

The number of times a call rings at a particular coverage point before the switch moves the call to the next coverage point depends on the type of ringing coverage point (for example, local, Distributed Communications System (DCS), CCRON, and so forth). For each type of coverage point, the following table shows which subsequent redirection interval on the System-Parameters Call Coverage/Call Forwarding screen is used.

Type of Coverage	Subsequent Redirection Interval
local	local
remote	*
CCRON	off-net
DCS	local

- Local — On the System-Parameters Coverage/Forwarding screen, the Local Subsequent Redirection/CFWD Don’t Answer Interval field.
- Off-net — On the System-Parameters Coverage/Forwarding screen, the Offnet Subsequent Redirection/CFWD Don’t Answer Interval field.
- * — The call is left off-net.

Call redirection criteria

Redirection criteria determine the conditions under which a call redirects from the principal (called) extension to the first position in the coverage path. The criteria and conditions that apply are as follows:

- Active

  Redirects calls to coverage immediately when the principal is active on at least one call appearance. For a phone with only one appearance or a single-line extension, assign the Busy criterion (discussed below) instead of the Active criterion.

- Busy

  Redirects calls to coverage when all available call appearances at the principal extension are in use. For multiappearance phones, one call appearance can be reserved for outgoing calls or incoming priority calls (discussed later). The remaining assigned call appearances are available for other incoming calls. An incoming call (other than a priority call) redirects to coverage only when all of these unreserved call appearances are in use. If at least one unreserved call appearance is idle at the principal extension, the call remains at that idle appearance.

  A Terminating Extension Group (TEG) is considered busy if any phone in the group is active on a call.

  Each phone in a UCD or DDC group must be active on at least one call appearance for the call to redirect to coverage. If any phone in the group is idle the call directs to that phone. If no phone is available, the call can queue if queuing is provided. If queuing is not provided, then the call routes to coverage. If the queue is full or all agents are in an auxiliary state, the group is considered busy and the call routes to coverage. Queued calls remain in queue for the Don’t Answer Interval.

  A call will not cover to a hunt group if no agents are logged in, or if all agents are in AuxWork mode.

- Don’t Answer

  Redirects calls to coverage if unanswered during the assigned Don’t Answer Interval. A call rings for the Don’t Answer Interval and then redirects to coverage.

- Cover All Calls

  Redirects all incoming calls to coverage. This criterion has precedence over any other criterion previously assigned.

- Send All Calls/Go to Cover

  Allows users to activate Send All Calls or Go to Cover as an overriding coverage criteria. This redirection criteria must be assigned before a user can activate Send All Calls or Go to Cover (discussed later).

- No Coverage

  Occurs when none of the above criteria are assigned. Calls redirect to coverage only when the principal has activated Send All Calls or the caller has activated Go to Cover. Both of these overriding criteria are discussed later.

Redirection criteria can be assigned in combinations. For example, you can combine Active/Don’t Answer and Busy/Don’t Answer. Other combinations are not possible or do not provide any useful function. For example, Active/Busy does not accomplish anything. A busy phone is always active.

Redirection criteria are assigned separately for internal and external calls. By linking the coverage paths, Busy/Don’t Answer can be assigned for internal calls and Active can be assigned for external calls. Similarly, Busy/Don’t Answer can apply for external calls and No Coverage can apply for internal calls. In the latter case, internal calls remain directed to the called phone or group.

All calls extended by the attendant are treated as external.

Warning users if their calls are redirected

You can warn analog phone users if they have features active that may redirect calls. For example, if the user has activated send all calls or call forwarding, you can administer a setting to play a special dial tone when the user goes off-hook. See ‘‘Special Dial Tone’’ on page 791 for more information.

Features that override Call Coverage

Some system features override Call Coverage criteria; they are checked before the redirection criteria are checked. These features are:

- Call Forwarding All Calls

  Call Forwarding provides a temporary override of the redirection criteria, if Send All Calls is not active. The call attempts to complete to the forwarded-to extension before redirecting to coverage. If the principal’s redirection criteria are met at the forwarded-to extension, the call redirects to the principal’s coverage path.

- Go to Cover

  Go to Cover allows users, when calling to another internal extension, to send the call directly to coverage. This is optionally assigned to a button on a phone and is activated by the internal calling party. Use of Go to Cover is discussed later.

- Send All Calls

  Send All Calls allows principals to temporarily direct all incoming calls to coverage regardless of the assigned redirection criteria. For example, if the redirection criteria are administered so that no calls redirect, all incoming calls terminate at the principal's phone unless Send All Calls is activated. Also, Send All Calls allows covering users to temporarily remove their phones from another user's coverage path.

  Send All Calls is activated by pressing the Send All Calls button or by dialing the Send All Calls access code. It is deactivated by pressing the button a second time or by dialing the deactivate code.

  A user who is not assigned a coverage path with Send All Calls or Cover All Calls redirection criteria, cannot activate Send All Calls.

  Any attempt to activate Send All Calls is denied if the currently active coverage path does not allow it in its coverage criteria. However, if the user activates Send All Calls for a coverage path that does allow it, and then the user's coverage path is changed by the system to a coverage path that does *not* allow Send All Calls:

  — The Send All Calls button remains lit.

  — Send All Calls automatically resumes when the user is changed back to a coverage path that does allow it.

  If a user has activated Send All Calls and has only one coverage point, and receives a call from that coverage point, the call rings silently at the user's phone, because the coverage point is already on the call.

  Send All Calls is similar to Cover All Calls, discussed previously. However, you set Cover All Calls and it is used for screening the principal's call. The principal may or may not be rung on an incoming call, depending on how this function is assigned. Send All Calls is controlled by the principal and is normally used when the principal is away temporarily.

  TEG calls are not affected by Send All Calls.

- Send Term

  Send Term is the Send All Calls equivalent for TEG. Since a TEG cannot be in a coverage path, Send Term applies only to a directly called TEG.

Conditions that override Call Coverage

Call Coverage provides redirection of calls from the called principal or group to alternate answering positions when certain criteria are met. Certain provisions allow calls to direct to and/or be answered by the principal even though the redirection or overriding criteria are met. These provisions are:

- If no answering positions are available in the Coverage Path, the call rings the called phone, if possible; otherwise, the calling party receives busy tone. This applies even if the Cover All Calls redirection criterion or the Send All Calls overriding criterion is active.
- Similarly, calls directed to a UCD or DDC group are queued, if queuing is available, when no group members are available to answer the call. The call remains in queue for the Don’t Answer Interval before routing according to the coverage path. If no points on the path are available, the call remains in queue. The worst case is when group queuing and the coverage points both are unavailable. In this case, the caller receives busy tone or ringback, depending on the type of trunk carrying the call.
- If the redirection criterion is Active or Cover All Calls, a called principal can receive a redirection notification signal (a short burst of ringing) when the call routes to coverage. (Redirection Notification is optional on a per-phone basis.) Note that in the Active, Cover All Calls, and Don’t Answer cases, the principal could answer the call. Busy means no call appearances are available to answer the call. Redirected calls maintain an appearance on the called phone, if possible. The call appearance status lamp flashes to indicate an incoming call before the call redirects. When the call does redirect, the status lamp continues to flash (when redirecting to AUDIX, the lamp goes out). The user can answer the call by pressing the call appearance button. If the call has already been answered by a covering user, the principal may bridge onto the call. This provision is called Simulated Bridged Appearance. If a covering user answers the call, the status lamp on the principal’s phone lights steadily.
- A phone user can use Directed Call Pickup to pick up a principal’s call or a call alerting at a coverage point. Directed Call Pickup allows a phone user to answer an alerting call from any station on the Avaya MultiVantage. That is, the alerting and answering stations do not need to be members of the same Call Pickup group. You enable and disable Directed Call Pickup on a system-wide base. However, permission to use the feature can be allowed or disallowed based on COR.
- Priority Calling, Dial Intercom, and Automatic Intercom Calls always route directly to the principal’s phone until the calling party activates Go to Cover. These calls take precedence over the redirection criteria and can seize the call appearance normally reserved for outgoing calls, if no other call appearances are available.

An internal calling party is informed that a call is redirecting to coverage by a single, short burst of ringing, called a Call Coverage tone. This tone is followed by an optional period of silence, called a Caller Response Interval (administered on the System-Parameters Call Coverage/Call Forwarding screen). This interval allows the calling party time to decide what to do: hang up or activate Leave Word Calling, Automatic Callback, or Go to Cover. Activating Go to Cover cancels the remaining interval.

Covering-user options

For specific Call Coverage needs, the following options are available to phone users:

- Consult

  Allows the covering user, by first pressing the Transfer button and then the Consult button, to call the principal (called party) for private consultation. These two actions place the caller on hold and establish a connection between the principal and the covering user. If the principal wishes, the covering user can complete the conference and add the calling party to the conversation. Similarly, the call can be transferred to the principal. Consult calls use the Simulated Bridged Appearance maintained on the call, if there is one. If not, the Consult call seizes any idle call appearance. If there is no idle call appearance, the Consult call is denied.

- Coverage Callback

  Allows a covering user, by pressing the Cover Callback button, to leave a message for the principal to call the calling party. Coverage Callback uses Implied Principal Addressing to infer both extensions so that the covering user does not have to dial either the principal’s or the caller’s number. The caller must be an internal caller. The principal receives no indication that the covering user handled the call.

  Alternatively, if the covering user presses the Leave Word Calling button, a “call me” message is left for the principal. The principal calls the covering user to get the message. This method is used when an external call is received or when an internal caller wants to leave a message but is not available for a return call.

- Coverage Answer Group

  A Coverage Answer Group can have up to eight members. When a call is redirected to a Coverage Answer Group, all phones in the group ring simultaneously. Anyone in the group can answer the call. Note that a bridged appearance of a coverage answer group member does not ring when calls cover to the group. A Coverage Answer Group member already handling a group call is rung when another call is redirected to that Coverage Answer Group. If a Coverage Answer Group member is also a member of another Coverage Answer Group, he or she can also receive calls for the other group. A second call directed to a Coverage Answer Group lights a Coverage Incoming Call Identification (ICI) lamp, if administered.

- Coverage Incoming Call Identification

  A Coverage ICI button can be assigned to multiappearance phone users without a display in a Coverage Answer Group.

  The Coverage ICI status lamp identifies a call incoming to that Coverage Answer Group. If a Coverage Answer Group is assigned to more than one Call Coverage path, the path number cannot be identified. Likewise, if a given path is assigned to more than one principal, the individual principals cannot be identified. To provide unique path and principal identification, you must establish a unique path for each principal and a unique Coverage Answer Group to be included in the path. A second coverage call takes control of the Coverage ICI lamp and does not return control to the previous call when the second call is released.

Time-of-Day Coverage

The Time-of-Day Coverage Table allows you to redirect calls to different lead-coverage paths at different times of the day and on different days of the week.

For example, an employee may want incoming calls to cover to a co-worker (office) during normal business hours, to cover to an off-net destination (home) in the early evening, and to cover to AUDIX at all other times. By specifying the appropriate lead-coverage paths in the Time-of-Day Coverage Table, the employee can have the call redirection flexibility shown in the following table. (If you were actually administering a Time-of-Day Coverage Table, you would provide the lead-coverage path numbers that redirect the calls to the employee's office, to their home, and to AUDIX.

Day of the Week	Time 1 Directed To	Time 2 Directed To	Time 3 Directed To	Time 4 Directed To
Monday	00:00 CovPath3 (AUDIX)	08:00 CovPath1 (Office)	17:30 CovPath2 (Home)	20:00 CovPath3 (AUDIX)
Tuesday	00:00 CovPath3 (AUDIX)	08:00 CovPath1 (Office)	17:30 CovPath2 (Home)	20:00 CovPath3 (AUDIX)
Wednesday	00:00 CovPath3 (AUDIX)	08:00 CovPath1 (Office)	17:30 CovPath2 (Home)	20:00 CovPath3 (AUDIX)
Thursday	00:00 CovPath3 (AUDIX)	08:00 CovPath1 (Office)	17:30 CovPath2 (Home)	20:00 CovPath3 (AUDIX)
Friday	00:00 CovPath3 (AUDIX)	08:00 CovPath1 (Office)	17:30 CovPath2 (Home)	
Saturday	00:00 CovPath3 (AUDIX)			
Sunday	00:00 CovPath3 (AUDIX)			

Time is represented in 24-hour format and activation times are ascending from the earliest to the latest. There are no gaps in the activation times; the entire day is covered. If you do not assign a lead-coverage path to a specific time interval, there is no coverage from that time until the next activation time with an assigned lead-coverage path.

When a call arrives at a principal, the system queries for the lead coverage path in effect at that time and uses that information to determine call redirection. If call coverage is changed via administration *while the call is in progress*, the administration changes do not affect that call.

Off-Net Call Coverage

Call Coverage allows a call to be redirected to a destination on the public network. The remote (off-net) number is administered on the Remote Call Coverage Table screen and may have up to 16 digits including either the outgoing trunk access code (TAC) or the feature access code (FAC) specifying ARS or AAR. Any coverage point can be an off-net destination.

Whenever an incoming trunk call is redirected off-net (coverage or forwarded), a timer is set that precludes any other incoming trunk call from redirecting off-net until the timer either expires or is cancelled. The rationale for this mechanism is to prevent calls that were redirected off-net from being re-routed back to the original principal from the off-net destination, effectively creating a round-robin loop that continuously seizes trunks until they are exhausted.

Avaya MultiVantage provides the means of performing call classification on an off-net coverage call to determine its disposition. If the off-net call is carried completely over ISDN facilities to its final destination, then ISDN trunk signaling is used to monitor the call. Otherwise, the system uses a call classifier port.

When the Avaya MultiVantage tries to classify an off-net coverage call (CCRON) using a call classifier port, the system introduces an unavoidable cut-through delay while the call classifier port attempts to identify an answered call. Neither the originating nor the answering party can hear each other during this delay of up to 1 second. A call classifier is attached to all off-net coverage calls made over analog facilities and also over ISDN facilities if the call is interworked to non-ISDN facilities on the public network.

When the Coverage of Calls Redirected Off-Net Enabled field on the System-Parameters Coverage-Forwarding screen is **y**:

- The system monitors off-net calls (call classification) and brings them back to the switch if they are not answered within the defined time interval. (Set this interval in the Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval field on the System-Parameters Coverage-Forwarding screen.) Calls also return to the switch if the system detects a call progress tone, such as busy or reorder.
- A simulated bridge appearance (SBA) is put on the principal and the green lamp is put in flashing mode; the principal can pick up the call at any time. You have the option of dispensing with call classification (and consequently the SBA) on a final CCRON coverage point.
- When a call classifier port is used to classify the call, the switch plays local ringback tone to the caller while the off-net call is being classified, concealing from the caller what is happening on the public network. When the call is answered off-net, it is likely that the first few syllables spoken by the answering party will not be heard by the calling party.
- If any party on the call is on HOLD when the call routes off net, the call classifier is removed from the call. The call behaves in the same manner as off-net calls when the Coverage of Calls Redirected Off-Net field is **n**.

- While an off-net call is undergoing call classification, any party who is not already on the call will be unable to bridge onto the call. Also, the originating party cannot release the call, conference anyone else onto the call, or transfer the call to a new party. Once the call is answered off-net or the call is returned to the switch for further call processing, then these restrictions are removed.
- If the last point in a coverage path is an off-net destination and no trunks are available to route the call, the switch attempts to re-terminate the call to the principal.
- Avaya MultiVantage has no control over any redirection of the call that may take place at an off-net destination. However, further coverage treatment will be provided if the Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval field expires before the call is answered off-net.

If the Coverage of Calls Redirected Off-Net field is not activated, the system does not monitor off-net calls and bring them back for additional call-coverage processing. In this case, once a call is directed to the remote call-coverage point, the principal is dropped from the call. Effectively, the off-net coverage point is the last coverage point in the coverage path.

VDN in a Call Coverage path

Assigning a VDN extension as the last point in a Call Coverage path allows the functionality of Call Vectoring to be applied to a coverage point. The programmable vector associated with the VDN effectively can provide great flexibility in call handling. The integration of the Call Vectoring and Call Coverage features can drive powerful AUDIX and Message Server applications.

For example, you can program the vector assigned to the VDN in the coverage path to queue a redirected call to a messaging split for call answer operation and to allow the caller to leave a message for the called principal. The same VDN also can be used to retrieve messages. The vector program may also be varied by time of day or split status to provide different types of coverage.

When a redirected call covers to a VDN, the principal’s simulated bridged appearance is removed when vector processing begins.

When covered or direct calls are connected to AUDIX or a messaging split via call vectoring, the original reason for redirection and the called principal must be passed to the adjunct over the Switch Communication Interface (SCI) link.

Use of a VDN as a coverage point provides integration to Centralized Messaging. That is, the Distributed Communications System (DCS) message sent to the remote switch with AUDIX includes the original reason for redirection and called principal.

An administration change is required to allow an extension that is assigned as a VDN to be entered as the last point in the coverage path. See *Avaya MultiVantage™ Call Center Software Release 11 Call Vectoring and Expert Agent Selection (EAS) Guide* for more information.

Extended User Administration of Redirected Calls

The Extended User Administration of Redirected Calls feature (also called telecommuting access) allows system users to change their lead-coverage path (or time of day table) or their call-forwarding designated destination from any on-site or off-site location. Refer to ''Extended User Administration of Redirected Calls'' on page 1554 for more information.

Measurements and reports

Three reports provide measurement information about Call Coverage:

- The Coverage Path Measurement Report describes coverage activity as it relates to the coverage paths.
- The Principal Coverage Measurement Report describes coverage activity as it relates to the principal extensions.
- The Call Detail Recording (CDR) shows the outgoing trunk calls.

For each report, a selection form lists the specific coverage paths or principal extensions to be measured. For more detailed information on these reports and their associated commands, refer to *Reports for Avaya MultiVantage™ Software*.

Considerations

- Incoming tie-trunk calls can be administered as either internal or external and are redirected to Call Coverage accordingly.

Guidelines and examples of Call Coverage

Call Coverage is an extremely flexible feature and allows several combinations of coverage points. To illustrate the usefulness of Call Coverage, three typical coverage arrangements are given below.

- Executive Coverage

  Provides a principal with call redirection to covering users having a close working relationship with the principal. Because of the status of the principal, personalized answering should be provided. Also, the principal may or may not choose to answer his or her own calls.

  Redirection of a principal's calls to a secretary is a typical example of this form of coverage. The secretary would be informed of the principal's daily schedule and other useful information such as the importance of certain calls. The secretary could provide personalized answering by answering calls with the principal's name.

  If the secretary is unavailable to answer the coverage call for the principal, the call redirects to a backup answering position. Personalized answering should also be provided at the backup position.

- Middle Manager Coverage

  Provides a group of principals with call redirection to one or more covering users (such as a secretary). The secretary should have some knowledge of the principal's daily schedule. A backup answering position should be provided in case the secretary is unavailable.

- General User Coverage

  Provides less-personal coverage for a broader spectrum of users. Covering users typically consist of a group or pooled answering arrangement. Coverage calls may be distributed among the members of the answering group.

Following is an example of how to provide an Executive Coverage arrangement.

1. Establish a unique Call Coverage Path for the executive.

   — If the secretary screens calls, specify Cover All Calls as the redirection criteria.

   — If the executive answers calls, specify Active, Busy, Don't Answer, Active/Don't Answer, or Busy/Don't Answer as desired.

   — Specify the secretary and the backup position (or the Coverage Answer Groups containing the secretary's and backup position's extensions) as the coverage points in the path.

2. If a Coverage Answer Group has been chosen as a coverage point, expect the following behavior:
   - — Note that, if the secretary and/or backup answering position are in a Coverage Answer Group, each receives only one redirected call for the executive at any given time.
   - — Only one call may cover and ring at a time.
   - — Calls do ring a Coverage Answer Group member already busy on a call to the group.
3. Optionally, specify a Send All Calls button on the executive's phone. If someone else answers the executive's calls via bridging, the button is not needed.
4. Specify a Send All Calls button and a Consult button on the secretary's phone. Specify a Coverage ICI button if the secretary does not have a call display. Send All Calls is needed if the secretary is unavailable for a period of time. Consult is needed to enable private consultation with the executive during an established call. Coverage ICI is needed to identify the call as a call to the executive rather than a personal call to the secretary.
5. Specify a Consult button and a Coverage ICI button on the backup position's phone for the same reasons that these buttons were specified for the secretary.

Interactions

- Agent Call Handling

  Do not assign Cover All Calls to agents with the Automatic Answer option. Any call (ACD or non-ACD), to an extension that has Automatic Answer enabled and has its coverage redirection criteria administered as Cover All Calls does not go to coverage but to the called extension. Cover All Calls redirection criteria have no effect on an incoming call when a user is in the Auto-Answer mode.

- Answer Detection

  Coverage of Calls Redirected Off-Net competes with Answer Detection for call classifier ports.

- Automatic Callback and Ringback Queuing

  Callback calls do not redirect to coverage. The caller can activate Automatic Callback when ringing, redirection notification signal, or busy signal is heard.

- Automatic Intercom, Dial Intercom, and Priority Calling

  Calls using these features are not redirected to coverage unless the caller presses the Go to Cover button.

- Bridged Call Appearance

  Coverage criteria for bridged call appearances are based entirely on the criteria of the primary extension associated with the bridged call appearance.

  If a phone user has activated Send All Calls on the primary extension, incoming calls still ring bridged call appearances of that extension as long as a simulated bridged appearance of the call is maintained at the primary extension.

  The switch blocks a user from bridging onto a call that has routed off-net while the call is undergoing call classification.

- Call Detail Recording

  When the Coverage of Calls Redirected Off-Net field is y, a CDR record is generated only after the call has actually been answered off-net. The dialed number in the record is the off-net number to which the call covers. The calling number is the station that is covered to the off-net location.

- Call Forwarding

  Call Forwarding provides a temporary override of the redirection criteria. Normally, calls forward instead of redirecting to coverage. When a forwarding extension’s redirection criteria are met at the designated (forwarded-to) extension, the call redirects to the forwarding extension’s coverage path.

  The system allows calls forwarded off-net to be tracked for busy or no-answer conditions and to return for further call-coverage processing under those conditions. However, if the principal does not have a coverage path, the system does not track the call and it is left at the off-net destination regardless of whether it is answered or busy.

  If an extension has both Send All Calls and Call Forwarding activated, most calls to that extension are immediately redirected to coverage. However, Priority Calls, are forwarded to the designated forwarding destination.

  If Cover All Calls is part of the coverage redirection criteria and if Call Forwarding is active at an extension, most calls to that extension are immediately redirected to coverage. However, Priority Calls, are forwarded to the designated forwarding destination.

  Activation of Send All Calls at the forwarded-to extension does not affect calls forwarded to that extension.

- Call Pickup

  Any call redirected to a covering user who is a member of a Call Pickup group can be answered by other members of the group.

- Call Prompting

  Coverage of Calls Redirected Off-Net competes with the Call Prompting feature for call classifier ports.

- CallVisor ASAI

  Coverage of Calls Redirected Off-Net competes with CallVisor for call classifier ports.

- Centralized Attendant Service

  If an incoming CAS call is directed to a hunt group, the call is not redirected to the hunt group's coverage path.

- Class of Restriction and Controlled Restrictions

  Users who normally may be restricted from receiving calls still can receive calls directed to them via Call Coverage.

- Conference

  The switch blocks users from conferencing another party onto a call which has routed off-net while that call is undergoing call classification. If any party on the call is on hold, the call routes off-net, but without undergoing call classification, even when the Coverage of Calls Redirected Off-Net field is **y**.

  A call that covers to a VDN cannot be added to a conference while the call is in vector processing. For example, say user A calls user B. B wants to have a three-way conference call and calls C. C doesn't answer, the call covers to a VDN, and from there enters a vector. Until vector processing has completed this call to some destination, the conference cannot be established.

- Direct Department Calling, Uniform Call Distribution, and Automatic Call Distribution

  If a user has an Auxiliary Work button, and activates or deactivates Send All Calls, the Auxiliary Work function associated with DDC or UCD is activated or deactivated simultaneously.

  If a user has no Auxiliary Work button, activating or deactivating Send All Calls still makes the user unavailable or available, respectively, for DDC and UCD calls, but Auxiliary Work is not activated or deactivated. The Auxiliary Work mode may be activated or deactivated using a feature access code.

  Activating or deactivating the Auxiliary Work function does not activate or deactivate Send All Calls.

- Direct Outward Dialing (DOD)

  Coverage of Calls Redirected Off-Net competes with DOD for call classifier ports when DOD uses Multi-Frequency Compelled (MFC) signaling. The Call Classifier - Detector port provides the MFC tones. Non-MFC DOD calls do not need the Call Classifier - Detector port for this purpose.

- Global Call Classification

  To classify tones in countries not using the USA tone plan, time cadences and frequencies must be administered so they can be downloaded to the call classification circuit packs. You need a Call Classifier - Detector or Tone Clock with Call Classifier - Tone Detector circuit pack.

- Hold

  If a covering user puts a call on hold, and the principal picks up on the call, the coverage appearance may or may not be dropped, depending on administration.

  If any party is on hold when a coverage call routes off-net, that call does not undergo call classification, even when the Coverage of Calls Redirected Off-Net field is **y**.

- Internal Automatic Answer

  If an internal call is redirected to another phone by a Call Coverage redirection criteria, then that call is eligible for IAA at that phone.

  IAA does not apply to calls to the original called extension when:

  — The called phone has activated Do Not Disturb, Send All Calls, or Cover All Calls

  — The calling phone has selected Go To Cover before placing the call

  Calls directed to a Coverage Answering Group are not eligible for IAA.

- ISDN End-to-End Calls

  When ISDN facilities carry an off-net coverage call entirely (end-to-end), call classification is accomplished through the ISDN protocol rather than by a call classifier port.

- Leave Word Calling

  Call Coverage can be used with or without LWC. However, the two features complement each other. When a covering user activates LWC during a coverage call, a message is left for the principal to call the covering user. When a covering user activates Coverage Callback during a coverage call, a message is left for the principal to call the internal caller.

- Night Service

  Calls routed to the night station via Night Service follow the coverage path of the night extension under all coverage criteria except Send All Calls.

- Privacy — Manual Exclusion

  When the primary or principal user bridges onto a call that went to coverage and has been answered at the coverage point, the user is not dropped when Privacy — Manual Exclusion is activated.

- R2-MFC Signaling

  Coverage of Calls Redirected Off-Net competes with the R2-MFC Signaling feature for Call Classifier - Detector ports.

- Simulated Bridged Appearance

  Calls redirected to coverage maintain an appearance on the called phone if a call appearance is available to handle the call. The called party can bridge onto the call at any time. The system can be administered to allow a simulated bridged appearance of the call to either remain at or be removed from the covering phone after the principal bridges onto the call.

  A simulated bridged appearance is maintained for calls covered by an off-net coverage point if the Coverage of Calls Redirected Off-Net feature is enabled. A simulated bridged appearance cannot be maintained for calls if the coverage point is linked to AUDIX.

  Consult calls use the simulated bridged appearance maintained on the call. At the conclusion of a consult call, the bridged appearance is no longer maintained. If the principal chooses not to talk with the calling party, the principal cannot bridge onto the call later.

  If a call that has, or has had, a simulated bridged appearance is conferenced or transferred, and redirects to coverage again, a simulated bridged appearance is not maintained at the conferenced-to or transferred-to extension.

- Tenant Partitioning

  The caller and called party must be able to access a coverage point. The caller is considered to be the covering user and the called party is considered to be the covered user. Both parties must be able to access the coverage point.

- Transfer

  The switch blocks a user from transferring a call which has routed off-net to another party while that call is undergoing call classification. If any party on a call that has routed off-net is on hold, the call does route off-net without undergoing call classification, even when the Coverage of Calls Redirected Off-Net field is **y**.

Transfers with Call Coverage as listed in the following table.

Source	Transfer initiator	Destination	Coverage type
External	Local station	Local station	External
	Local station	Remote station	External
	Remote station	Local station	Internal
	Remote station	Remote station	Internal
	Attendant	Local station	External
	Attendant	Remote station	External
Internal	Local station	Local station	Internal
	Local station	Remote station	Internal
	Remote station	Local station	Internal
	Remote station	Remote station	Internal
	Attendant	Local station	External
	Attendant	Remote station	External

NOTE:
The coverage criteria for transferred DID calls depends upon the External Treatment For Transferred Incoming Calls field on the Feature-Related System Parameters screen.

Related topics

Refer to ''Coverage Answer Group'' on page 670 for information about and field descriptions on the Coverage Answer Group screen.

Refer to ''Coverage Path'' on page 672 for information about and field descriptions on the Coverage Path screen.

Refer to ''Feature Access Code'' on page 752 for information on activating or deactivating sending calls to coverage.

Refer to ''System Parameters Call Coverage / Call Forwarding'' on page 1104 for information on setting the system-wide parameters for call coverage and call forwarding.

Refer to ''Hunt Group'' on page 840 for information on sending calls to a hunt group extension.

Refer to ''Remote Call Coverage Table'' on page 1037 for information about and field descriptions on the Remote Call Coverage Table screen.

Refer to ''Station'' on page 1066 for information on assigning feature buttons.

Refer to ‘‘Terminating Extension Group’’ on page 1156 for information on assigning a coverage path to a Terminating Extension Group.

Refer to ‘‘Time of Day Coverage Table’’ on page 1161 for information about and field descriptions on the Time of Day Coverage Table screen.

Refer to ‘‘Trunk Group’’ on page 1167 for information on specifying internal ringing and coverage for incoming calls.

Refer to ‘‘Setting up basic call coverage’’ on page 187 for instructions on administering basic call coverage.

Refer to ‘‘Setting up advanced call coverage’’ on page 191 for instructions on administering advanced call coverage.

Call Detail Recording

CDR tracks call information on a per-trunk-group or station-to-station basis. For every trunk group (including auxiliary trunks) that you administer for CDR reports, the system keeps track of incoming, outgoing and tandem calls. You can also receive reports on temporary signaling connections (TSCs) that involve trunks, and calls made using loudspeaker paging or code calling access.

Detailed description

You can also ask the system to report on ineffective call attempts. This may greatly increase the number of calls recorded, but may also help you to increase security, since the system records call attempts that are blocked because of insufficient calling privileges. This can also let you know if your users are not able to make calls because all trunks are busy.

CDR tracks the audio service link calls that the switch uses for IP softphones set up as telecommuter. An IP softphone may use one audio service link to make many short calls, but this appears as one long call on the CDR reports.

Some call accounting systems do not support all of the information offered by CDR. See your Avaya representative for details.

Account Code Dialing

Account Code Dialing allows certain calls to be associated with a particular project or account number. To do this, users enter account codes when they place certain types of outgoing calls. These account codes then appear on the call record, which you can use for accounting or billing purposes. Account code dialing can be optional or mandatory (forced).

To associate an account code with a particular call, a user first dials the CDR account code access code that you have established, then dials the desired account code. Next, the user dials the desired trunk access code, or ARS access code, followed by the phone number.

The call record contains the account code, TAC or ARS access code, and the phone number. It does not contain the account code FAC.

Forced Entry of Account Codes

You can force your users to enter account codes on a system-wide, per-trunk or per-user basis, or on the basis of the digit string the user dials. If you have this feature in place, the system rejects any call that requires an account code where one is not supplied. To maximize system security, it is recommended that you use Forced Entry of Account Codes (FEAC).

SECURITY ALERT:

Avaya MultiVantage does not verify account codes. As long as the user enters a digit string of the appropriate length, the system allows the call. You must use Authorization Codes if you want the system to determine if the user is authorized to make the call.

You can assign FEAC in the following ways:

- All calls marked for FEAC on the Toll Analysis table

  If you activate this CDR system parameter, all users must dial an account code when the digits dialed match a digit string with FEAC=y. This includes calls made by ARS or TAC.

- Toll calls made by users with a specific class of restriction (COR)

  If FEAC is assigned to a specific COR, any telephone user assigned that COR must dial an account code before making calls that are administered forced entry of account codes.

- All calls made over a trunk group with FEAC in COR (TAC calls)

  Users cannot access a trunk group that is assigned a COR with FEAC until they dial an account code. If a call is routed via ARS, the system does not check for FEAC in the trunk group's COR. Therefore, if you want your users to enter account codes for ARS calls, you must administer this in the Toll Analysis table.

If an account code is required for a call and the user does not enter one, the system responds with intercept tone.

The following types of calls never require an account code:

- Attendant originated call
- Busy verification of a trunk by an attendant or telephone user
- DCS (unless required by the trunk group's COR)
- Personal CO Line
- Remote access without barrier codes
- Trunk-to-trunk connections

Call Splitting

Call splitting keeps track of calls where more than two parties are involved. These can be calls that are transferred, conferenced, or where an attendant becomes involved. If you have call splitting activated and any of these situations arise, CDR produces a separate record for each new party on the call.

You can administer call splitting for both incoming and outgoing trunks, and both can have attendant calls recorded separately.

Incoming trunk call splitting

If incoming trunk call splitting is enabled, CDR starts a new record whenever an incoming trunk call is conferenced or transferred. Whenever a user drops from the call or the call is successfully transferred, CDR produces a record relevant to this user's participation. These call records show the amount of time each party was on the call, the incoming trunk access code, the dialed number and the condition code, as well as the other fields specified in the record format.

For incoming trunk calls that are conferenced, CDR creates a new record whenever a new party comes on the call. The duration field in these records shows how long each party participated on the call. Conference calls produce records with duration fields that overlap. The duration of a transferred call begins when the transferring party presses the TRANSFER button for the second time, so there is no overlap.

If ITCS is enabled and an incoming trunk call is conferenced or transferred to a local extension that is optioned for Intraswitch CDR, the call produces an incoming trunk call record. It does not produce an Intraswitch record.

To enable this option, set the Incoming Call Splitting field to **y** on the CDR System Parameters screen, and make sure that Record Outgoing Calls Only = n.

ITCS examples

The following scenarios depict calls made with ITCS active. The tables that follow do not show all fields, only those that may change due to call splitting. Call durations are approximate.

Caller A (TAC 123) makes an incoming trunk call to switch party B (5657890). They talk for 2 minutes. B then conferences in C (ext. 54321), and D (ext. 59876). The entire group talks for another 8 minutes, at which point B drops off the call. This produces a record for segment A–B.

A, C and D continue to talk for another 5 minutes. All remaining parties drop, producing two more records; A–C and A–D. Note that each record shows the incoming trunk ID as the calling number.

Table 28. ITCS conference

Segment	Duration	Condition Code	Access Code Used	Calling Number	Dialed Number
A–B	0:10:0	C		123	5657890
A–C	0:13:0	C		123	54321
A–D	0:13:0	C		123	59876

A (TAC 123) calls Station B (5657890). They talk for 1 minute, then B transfers the call to C (54321). CDR generates a record for segment A–B. A and C talk for 5 minutes. CDR generates a record for segment A–C.

Table 29. ITCS transfer

Segment	Duration	Condition Code	Access Code Used	Calling Number	Dialed Number
A–B	0:01:0	9		123	5657890
A–C	0:05:0	9		123	54321

A (TAC 123) calls switch party B (5657890), they talk for one minute. B transfers the call to public-network party C (5665555), they talk for 4 minutes. Note that the duration of the original incoming trunk call includes the time after the call was transferred to an outgoing trunk, until all trunk parties drop.

Table 30. ITCS transfer to outgoing trunk

Segment	Duration	Condition Code	Access Code Used	Calling Number	Dialed Number
A–B	0:05:0	9		123	5657890
A–C	0:04:0	9	345	123	5665555

Outgoing trunk call splitting

If outgoing trunk call splitting is active, CDR creates records of transferred outgoing calls as described above for incoming trunk call splitting. For conferenced calls, the originator of the conference will be charged until he or she drops from the call, at which point CDR begins a second record for the conferenced user. Records for parties on a conference do not overlap; they are split. To enable this option, set the Outgoing Call Splitting field to **y** on the CDR System Parameters screen.

OTCS examples

In the next example, switch party A (57890) calls B(7771234), talks for 5 minutes, then conferences in C (7775678). They all talk for another 5 minutes, at which point all parties drop.

Table 31. OTCS conference

Segment	Duration	Condition Code	Access Code Used	Calling Number	Dialed Number
A–B	0:10:0	C	345	57890	7771234
A–C	0:05:0	C	345	57890	7775678

Switch party A (51234) calls public-network party B (5659999). They talk for 5 minutes. A then transfers the call to switch party C (54444).

Table 32. OTCS transfer

Segment	Duration	Condition Code	Access Code Used	Calling Number	Dialed Number
A–B	0:01:0	A	345	51234	5659999
C–B	0:05:0	A	345	54444	5659999

Attendant call recording

Both incoming and outgoing call splitting give you the option of recording the attendant portion of calls that are transferred. To enable this option, set the Incoming and/or Outgoing Attendant Call Record field to **y**.

If either incoming or outgoing trunk call splitting is enabled, the attendant portion of a conference call always produces a separate record.

Attendant call recording examples

Public-network party A (TAC 123) calls the attendant (Attd), and asks to be transferred to switch party B (58888).

Table 33. Attendant transfer incoming trunk

Segment	Duration	Condition Code	Access Code Used	Calling Number	Dialed Number
A–Attd	0:01:0	9		123	Attd
A–B	0:05:0	9		123	58888

The attendant (Attd) dials switch party A (59999), then transfers the call to public-network party B (4445678).

Table 34. Attendant transfer outgoing trunk

Segment	Duration	Condition Code	Access Code Used	Calling Number	Dialed Number
Attd–B	0:01:0	A	345	Attd	4445678
A–B	0:05:0	A	345	59999	4445678

Intraswitch CDR

Intraswitch CDR generates call records for calls to and from users on the local switch. For the system to generate an intraswitch CDR record, one of the extensions involved in a call must have intraswitch CDR assigned.

If a station is optioned for Intraswitch CDR, and ITCS is also enabled, ITCS overrides Intraswitch CDR. That is, incoming trunk calls involving the station produce trunk call records, not Intraswitch CDR records.

The output for intraswitch CDR follows the same format you have established for other call records. Certain fields do not appear on intraswitch call records, because they do not pertain to internal calls. For example, an intraswitch record does not contain trunk access codes or circuit IDs, since these do not apply.

Some calls may seem to be intraswitch CDR calls, but actually result in trunk calls. For example, a station-to-station call to an extension that is forwarded to an outgoing trunk produces only a trunk CDR record, regardless of whether or not either station has intraswitch CDR assigned.

You can assign intraswitch CDR to terminating extension groups (TEGs), stations, data modules, VDNs, PRI endpoints, access endpoints, or hunt groups. The number that appears in the dialed number field depends on whether you have administered CDR System Parameters to record hunt group/member or VDN information. You *cannot* assign intraswitch CDR to attendant consoles or CallVisor ASAI stations.

NOTE:
If an extension with intraswitch CDR is neither the originator of the call nor the dialed number of the call, the system does not produce a call record, even though the extension might be a party on the call (via Call Pickup, Call Forwarding, etc.).

Privacy

CDR Privacy allows you to administer the system to blank a given number of dialed digits from a CDR report. This is useful when it is necessary to know details of a person’s calls for accounting purposes, but it is not necessary or desirable to know the exact number called.

You can administer the number of digits to hide, up to 7. The value in Privacy Digits to Hide determines how many digits do not appear on the call record. This parameter is system-wide. Whether or not an individual’s calls receive Privacy treatment is determined by the CDR Privacy field on the Station screen.

When an adjunct-originated call is made on behalf of a hunt group and the CDR system parameter option is set to **group-ext**, then CDR privacy does not apply. If this field is set to **member-ext**, privacy does apply.

NOTE:
Certain countries have requirements that a certain number of digits must be blanked from every call. Also, certain report processors do not support this option.

CDR output

If your system uses two CDR output formats, one is administered as the primary CDR output format; the other is administered as the secondary CDR output format. The secondary output format is typically used for a local storage format (CDRU) to provide CDR data to NCOSS for assessing network performance or helping to find network problems.

The primary and secondary ports work independently. Each port will work even if the link to the other port is down. If a link is down for more than a minute, some data may be lost. However, the most recent 500 (Release 5vs/si/csi and later), or 1,900 (Release 5r and later) records are stored for the primary port even when a loss of records occurs. When the link comes back up, these records are output on a first-in, first-out basis.

If the CDR buffer is full, you can select a call record handling option to determine which of the following occurs:

- Calls are blocked with a reorder tone
- Overwrite old CDR records with new ones
- Calls are routed to an attendant as non-CDR calls

The following information applies to the port used for the secondary CDR output device:

- Data going to the secondary port should be the same as that going to the primary port. You can use the following record types for secondary output: LSU, Int-Direct, Int-Process, and Unformatted.
- If the system experiences problems in sending records to the primary CDR Output Device, the system discontinues sending records to the secondary port for 2 minutes. The secondary port should be run at the highest possible speed in order to prevent loss of information.
- If the output buffer is full, the system busies out the secondary port for 2 minutes. This makes system resources available to send data to the primary CDR port before the data is lost. The system continues to busy out the secondary port for 2-minute intervals until less than 400 records (1800 for Release 5r and later) remain in the buffer.

CDR Record formats

Two types of formats are sent to the CDR output device, date record and call detail formats.

Date record format

CDR sends date information to the CDR device once a day (at midnight), or when the device is connected. This is a non-call record, and contains only the information shown in the date record formats below.

Several formats are available for date records: one for CDRUs, one for the printer, and one for the TELESEER. The records sent to the TELESEER and printer contain the date only while the records sent to the CDRU contain time. See Table 35, Table 36, and Table 37.

NOTE:
The date can be in month/day *or* day/month format, as selected on the CDR System Parameters screen. The date/time may also be reversed for international standards.

Table 35. Date Record Format to LSU, LSU-Expand, Unformatted, and Customized

Position	Data Field Description
1–2	Hour (leading 0 added if needed)
3	Colon (:)
4–5	Minute (leading 0 added if needed)
6	Blank
7–8	Month (Leading Zero Added if Needed)
9	Slash (/)
10–11	Day (leading 0 added if needed)
12	Carriage return
13	Line feed
14–16	Null

Table 36. Date Record Format for Printer and Expanded

Position	Data Field Description
1–2	Month (leading 0 added if needed)
3	Space
4–5	Day (leading 0 added if needed)
6	Carriage return
7	Line feed
8–10	Null

Table 37. Date Record Format for TELESEER 59 Character, Int-Proc, Int-Direct, and Int-ISDN

Position	Data Field Description
1–2	Month (leading 0 added if needed)
3–4	Day
5	Carriage return
6	Line feed
7–9	Null

Call Detail Record formats

Customized format

You can use the customized record format to make up your own call record. You can determine the data elements you want and their positions in the record. This method may be necessary if you want to include certain data elements that are not available on the standard formats.

However, whatever device you use to interpret the CDR data needs to be programmed to accept these formats. Consult your Avaya representative before using a custom record format.

Table 38. CDR Data format — TELESEER

Position	Data Field Description
1–3	Space
4–5	Time of day-hours
6–7	Time of day-minutes
8	Duration-hours
9–10	Duration-minutes
11	Duration-tenths of minutes
12	Condition code
13–15	Access code dialed
16–18	Access code used
19–33	Dialed number
34–38	Calling number
39–53	Account code
54	FRL
55	IXC
56–58	Incoming circuit ID
59–61	Outgoing circuit ID
62	Feature flag
63–69	Authorization code
70–76	Space
77	Carriage return
78	Line feed
79–81	Null

Table 39. CDR Data Format — ISDN TELESEER

Position	Data Field Description
1–3	Space
4–5	Time of day-hours
6–7	Time of day-minutes
8	Duration-hours
9–10	Duration-minutes
11	Duration-tenths of minutes
12	Condition code
13–15	IXC
16–18	Access code used
19–33	Dialed number
34–38	Calling number
39–53	Account code
54	INS (units)
55	FRL
56–58	Incoming circuit ID
59–61	Outgoing circuit ID
62	Feature flag
63–69	Authorization code
70–71	INS (hundreds, tens)
72–76	Space
77	Line feed
78–80	Null

Table 40. CDR Data Format — Enhanced TELESEER

Position	Data Field Description
1–3	Space
4–5	Time of day-hours
6–7	Time of day-minutes
8	Duration-hours
9–10	Duration-minutes
11	Duration-tenths of minutes
12	Condition code
13–16	IXC code
17–19	Access code used
20–34	Dialed number
35–39	Calling number
40–54	Account code
55	ISDN NSV (units)
56	FRL
57–59	Incoming circuit ID
60–62	Outgoing circuit ID
63	Feature flag
64–70	Authorization code
71–72	ISDN NSV (hundreds, tens)
73–76	Space
77	Carriage return
78	Line feed
79–81	Null

Table 41. CDR Data Format — 59 Character

Position	Data Field Description
1–2	Time of day-hours
3–4	Time of day-minutes
5	Duration-hours
6–7	Duration-minutes
8	Duration-tenths of minutes
9	Condition code
10–12	Access code dialed
13–15	Access code used
16–30	Dialed number
31–35	Calling number
36–50	Account code
51	FRL
52	IXC
53–55	Incoming circuit ID
56–58	Outgoing circuit ID
59	Carriage return
60	Line feed
61–63	Null

Table 42. CDR Data Format — Printer

Position	Data Field Description
1–2	Time of day-hours
3–4	Time of day-minutes
5	Space
6	Duration-hours
7–8	Duration-minutes
9	Duration-tenths of minutes
10	Space
11	Condition code
12	Space
13–15	Access code dialed
16	Space
17–19	Access code used
20	Space
21–35	Dialed number
36	Space
37–41	Calling number
42	Space
43–57	Account code
58	Space
59–65	Authorization code
66–69	Space
70	FRL
71	Space
72	IXC
73	Space
74–76	Incoming circuit ID
77	Space

Continued on next page

Table 42. CDR Data Format — Printer (Continued)

Position	Data Field Description
78–80	Outgoing circuit ID
81	Space
82	Feature flag
83	Carriage return
84	Line feed

Table 43. CDR Data Format — ISDN Printer

Position	Data Field Description
1–2	Time of day-hours
3–4	Time of day-minutes
5	Space
6	Duration-hours
7–8	Duration-minutes
9	Duration-tenths of minutes
10	Space
11	Condition code
12	Space
13–15	IXC
16	Space
17–19	Access code used
20	Space
21–35	Dialed number
36	Space
37–41	Calling number
42	Space

Continued on next page

Table 43. CDR Data Format — ISDN Printer (Continued)

Position	Data Field Description
43–57	Account code
58	Space
59–65	Authorization code
66	Space
67–68	INS (hundreds, tens)
69	Space
70	INS (units)
71	Space
72	FRL
73	Space
74–76	Incoming circuit ID
77	Space
78–80	Outgoing circuit ID
81	Space
82	Feature flag
83	Carriage return
84	Line feed

Table 44. CDR Data Format — Enhanced Printer

Position	Data Field Description
1–2	Time of day-hours
3–4	Time of day-minutes
5	Space
6	Duration-hours
7–8	Duration-minutes

Continued on next page

Table 44. CDR Data Format — Enhanced Printer (Continued)

Position	Data Field Description
9	Duration-tenths of minutes
10	Space
11	Condition code
12	Space
13–16	IXC code
17	Space
18–21	Access code used
22	Space
23–37	Dialed number
38	Space
39–43	Calling number
44	Space
45–59	Account code
60	Space
61–67	Authorization code
68	Space
69–71	ISDN NSV
72	Space
73	FRL
74	Space
75–77	Incoming circuit ID
78	Space
79–81	Outgoing circuit ID
82	Space
83	Feature flag
84	Carriage return
85	Line feed

Table 45. CDR Data Format — LSU-Expand

Position	Data Field Description
1–2	Time of day-hours
3–4	Time of day-minutes
5	Space
6	Duration-hours
7–8	Duration-minutes
9	Duration-tenths of minutes
10	Space
11	Condition code
12	Space
13–15	Access code dialed
16–18	Access code used
19	Space
20–34	Dialed number
35	Space
36–39	Calling number
40	Space
41–45	Account code
46	Space
47–53	Authorization code
54–57	Space
58	FRL
59	Space
60	Calling number (1st digit)
61	Space
62–63	Incoming circuit ID (tens, units)
64	Space
65	Feature flag

Continued on next page

Table 45. CDR Data Format — LSU-Expand (Continued)

Position	Data Field Description
66	Space
67–68	Outgoing circuit ID (tens, units)
69	Space
70	Outgoing circuit ID (hundreds)
71	Space
72	Incoming circuit ID (hundreds)
73	IXC
74	Carriage return
75	Line feed
76–78	Null

Table 46. CDR Data Format — LSU

Position	Data Field Description
1	Duration-hours
2-3	Duration-minutes
4	Duration-tenths of minutes
5	Condition code
6–8	Access code dialed
9–11	Access code used
12–26	Dialed number
27–30	Calling number (digits 2–5 for 5-digit dial plan)
31–35	Account code (first 5 digits)
36–42	Authorization code or digits 6–12 of account code
43–44	Space or digits 13–14 of account code
45	FRL or digit 15 of account code

Continued on next page

Table 46. CDR Data Format — LSU (Continued)

Position	Data Field Description
46	Calling number (1st digit)
47–48	Incoming circuit ID (tens, units)
49	Feature flag
50–52	Outgoing circuit ID (tens, units, hundreds)
53	Incoming circuit ID (hundreds)
54	IXC
55	Carriage return
56	Line feed
57–59	Null

Table 47. CDR Data Format — ISDN LSU

Position	Data Field Description
1	Duration-hours
2–3	Duration-minutes
4	Duration-tenths of minutes
5	Condition code
6–8	IXC
9–11	Access code used
12–26	Dialed number
27–30	Calling number (digits 2–5 for 5-digit dial plan)
31–35	Account code (digits 1–5)
36–42	Authorization code or digits 6–12 of account code
43–44	INS or digits 13–14 of account code
45	INS (3rd digit), FRL, or digit 15 of account code
46	Calling number (1st digit of 5-digit calling number)

Continued on next page

Table 47. CDR Data Format — ISDN LSU (Continued)

Position	Data Field Description
47–48	Incoming circuit ID (tens, units)
49	Feature flag
50–52	Outgoing circuit ID (tens, units, hundreds)
53	Incoming circuit ID (hundreds)
54	FRL
55	Carriage return
56	Line feed
57–59	Null

Table 48. CDR Data Format — Enhanced LSU

Position	Data Field Description
1	Duration-hours
2–3	Duration-minutes
4	Duration-tenths of minutes
5	Condition code
6–9	IXC code
10–12	Access code used
13–27	Dialed number
28–31	Calling number
32–35	Account code (digits 1–4)
36–42	Authorization code or digits 6–12 of account code
43–45	ISDN NSV
46	1st digit of a 5-digit calling number
47–48	Incoming circuit ID (tens, units)

Continued on next page

Table 48. CDR Data Format — Enhanced LSU (Continued)

Position	Data Field Description
49	Feature flag
50–52	Outgoing circuit ID (tens, units, hundreds)
53	Incoming circuit ID (hundreds)
54	FRL
55	Carriage return
56	Line feed
57–59	Null

Table 49. CDR Data Format — Expanded

Position	Data Field Description
1–2, 3–4	Time of day-hours, -minutes
5	Space
6, 7–8, 9	Duration-hours, minutes, tenths of minute
10	Space
11	Condition code
12	Space
13–16	Access code dialed
17	Space
18–21	Access code used
22	Space
23–37	Dialed number
38	Space
39–48	Calling number
49	Space
50–64	Account code

Continued on next page

Table 49. CDR Data Format — Expanded (Continued)

Position	Data Field Description
65	Space
66–72	Authorization code
73-76	Space
77	FRL
78	Space
79–81	Incoming circuit ID
82	Space
83–85	Outgoing circuit ID
86	Space
87	Feature flag
88	Space
89–90	Attendant console
91	Space
92–95	Incoming trunk access code
96	Space
97–98	Node number
99	Space
100–102	INS
103	Space
104–106	IXC
107	Space
108	BCC
109	Space
110	MA-UUI
111	Space
112	Resource flag
113	Space

Continued on next page

Table 49. CDR Data Format — Expanded (Continued)

Position	Data Field Description
114–117	Packet count
118	Space
119	TSC flag
120	Space
121–129	Reserved
130	Space
131	Carriage return
132	Line feed
133–135	Null

Table 50. CDR Data Format — Enhanced Expanded

Position	Data Field Description
1–2	Time of day-hours
3–4	Time of day-minutes
5	Space
6	Duration-hours
7–8	Duration-minutes
9	Duration-tenths of minutes
10	Space
11	Condition code
12	Space
13–16	Access code dialed
17	Space
18–21	Access code used
22	Space

Continued on next page

Table 50. CDR Data Format — Enhanced Expanded (Continued)

Position	Data Field Description
23–37	Dialed number
38	Space
39–48	Calling number
49	Space
50–64	Account code
65	Space
66–72	Authorization code
73	Space
74–75	Time in queue
76	Space
77	FRL
78	Space
79–81	Incoming circuit ID
82	Space
83–85	Outgoing circuit ID
86	Space
87	Feature flag
88	Space
89–90	Attendant console
91	Space
92–95	Incoming TAC
96	Space
97–98	Node number
99	Space
100–102	ISDN NSV
103	Space
104–107	IXC code

Continued on next page

Table 50. CDR Data Format — Enhanced Expanded (Continued)

Position	Data Field Description
108	Space
109	BCC
110	Space
111	MA-UUI
112	Space
113	Resource flag
114	Space
115–118	Packet count
119	Space
120	TSC flag
121	Space
122–123	Bandwidth
124	Space
125–130	ISDN CC (digits 1–6)
131–135	ISDN CC (digits 7–11)/PPM count (1–5)
136–146	Reserved for future use
147	Carriage return
148	Line feed
149–151	Null

Table 51. CDR Data Format — Unformatted

Position	Data Field Description
1–2	Time of day-hours
3–4	Time of day-minutes
5	Duration-hours

Continued on next page

Table 51. CDR Data Format — Unformatted (Continued)

Position	Data Field Description
6–7	Duration-minutes
8	Duration-tenths of minutes
9	Condition code
10–13	Access code dialed
14–17	Access code used
18–32	Dialed number
33–42	Calling number
43–57	Account code
58–64	Authorization code
65–66	Space
67	FRL
68–70	Incoming circuit ID (hundreds, tens, units)
71–73	Outgoing circuit ID (hundreds, tens, units)
74	Feature flag
75–76	Attendant console
77–80	Incoming TAC
81–82	Node number
83–85	INS
86–88	IXC
89	BCC
90	MA-UUI
91	Resource flag
92–95	Packet count
96	TSC flag
97–100	Reserved

Continued on next page

Table 51. CDR Data Format — Unformatted (Continued)

Position	Data Field Description
101	Carriage return
102	Line feed
103–105	Null

Table 52. CDR Data Format — Enhanced Unformatted

Position	Data Field Description
1–2	Time of day-hours
3–4	Time of day-minutes
5	Duration-hours
6–7	Duration-minutes
8	Duration-tenths of minutes
9	Condition code
10–13	Access code dialed
14–17	Access code used
18–32	Dialed number
33–42	Calling number
43–57	Account code
58–64	Authorization code
65–66	Time in queue
67	FRL
68–70	Incoming circuit ID
71–73	Outgoing circuit ID
74	Feature flag
75–76	Attendant console number

Continued on next page

Table 52. CDR Data Format — Enhanced Unformatted (Continued)

Position	Data Field Description
77–80	Incoming TAC
81–82	Node number
83–87	ISDN NSV
88–89	IXC code
90	BCC
91	MA-UUI
92	Resource flag
93–96	Packet count
97	TSC flag
98–99	Bandwidth
100–105	ISDN CC (digits 1–6)
106–110	ISDN CC (digits 7–11)/PPM count (1–5)
111–114	Reserved for future use
115	Carriage return
116	Line feed
117–119	Null

Table 53. CDR Data Format — Int Process

Position	Data Field Description
1–2	Format code
3–4	Time of day-hours
5–6	Time of day-minutes
7	Duration-hours
8–9	Duration-minutes
10	Duration-tenths of minutes

Continued on next page

Table 53. CDR Data Format — Int Process (Continued)

Position	Data Field Description
11	Space
12	Condition code
13	Space
14–16	Access code dialed
17–19	Access code used
20	Space
21–38	Dialed number (digits 1–18)
39–43	Calling number (digits 1–5)
44	Space
45–59	Account code (digits 1–15)
60	Space
61	IXC
62	FRL
63–65	Space
66–67	Incoming circuit ID (digits 1–2)
68–70	Space
71–72	Outgoing circuit ID (digits 1–2)
73	Space
74–78	PPM count (digits 1–5)
79	Carriage return
80	Line feed
81–83	Null

Table 54. CDR Data Format — Int-Direct

Position	Data Field Description
1–2	Day of month
3–4	Month
5–6	Year
7	Space
8–9	Time of day-hours
10–11	Time of day-minutes
12	Space
13	Duration-hours
14–15	Duration-minutes
16	Duration-tenths of minutes
17	Space
18	Condition code
19	Space
20–22	Access code dialed
23–25	Access code used
26	Space
27–44	Dialed number used
45	Space
46–50	Calling number
51	Space
52–66	Account code
67	Space
68–72	PPM count
73	Space
74–75	Incoming circuit ID
76	Space

Continued on next page

Table 54. CDR Data Format — Int-Direct (Continued)

Position	Data Field Description
77–78	Outgoing circuit ID
79	Carriage return
80	Line feed

Table 55. CDR Data Format — Int-ISDN

Position	Data Field Description
1–2	Time of day-hours
3–4	Time of day-minutes
5	Space
6	Duration-hours
7–8	Duration-minutes
9	Duration-tenths of minutes
10	Space
11	Condition code
12	Space
13–16	Access code dialed
17	Space
18–21	Access code used
22	Space
23–37	Dialed number
38	Space
39–48	Calling number
49	Space
50–64	Account code

Continued on next page

Table 55. CDR Data Format — Int-ISDN (Continued)

Position	Data Field Description
65	Space
66–72	Authorization code
73	Space
74	Line feed
75	Space
76	FRL
77	Space
78	Incoming circuit ID (hundreds)
79	Incoming circuit ID (tens)
80	Incoming circuit ID (units)
81	Space
82–84	Outgoing circuit ID
85	Space
86	Feature flag
87	Space
88–89	Attendant console (1st digit)
90	Space
91–94	Incoming trunk access code
95	Space
96–97	Node number
98	Space
99–101	INS
102	Space
103–106	IXC
107	Space
108	BCC
109	Space

Continued on next page

Table 55. CDR Data Format — Int-ISDN (Continued)

Position	Data Field Description
110	MA-UUI
111	Space
112	Resource flag
113	Space
114–119	Reserved
120–124	PPM or reserved
125–131	Space
132	Carriage return
133	Line feed
134–136	Null

Call detail record field descriptions

The following list describes the CDR data collected for each call and the number of digits in each field. All information is right adjusted in the respective field, unless otherwise indicated. Where the field name for customized records is different from the standard, the custom field name appears in parentheses.

- **Access Code Dialed** (code-dial) — 3 or 4 digits

  The access code the user dialed to place an outgoing call. This can be the ARS access code, AAR access code, or the access code of a specific trunk group. This field is also used to record the X.25 Feature Access Code of an outgoing X.25-addressed call.

- **Access Code Used** (code-used) — 3 or 4 digits

  This field is used only for outgoing calls when the trunk group used is different from the access code dialed. It is not used when a TAC is dialed. For example, your system may use a feature access code for ARS. This field contains the access code of the actual trunk group that the call was routed over. When the dialed and used access code are the same, this field will be blank.

  If you use ISDN or enhanced formats with TELESEER, LSU, or Printer record types, this field always shows the access code of the trunk group, even if it is the same as the access code dialed.

- **Account Code** (acct-code) — up to 15 digits

  This field may contain a number to associate call information with projects or account numbers. For some formats, a long account code overwrites spaces on the record that are assigned to other fields.

- **Attendant Console** (attd-console) — 2 digits

  This field contains the attendant console number of the attendant that handled the call in a record that is marked as being attendant handled.

- **Authorization Code** (auth-code) — 4–13 digits

  This field contains the authorization code used to make the call. For all formats except the custom format, codes longer than 7 digits are truncated, keeping only the first 7 digits. For non-ISDN and ISDN LSU formats, the authorization code is fewer than 6 digits in length. It is 5 for Enhanced LSU. On the 59-character record, the authorization code is never recorded.

- **Bandwidth** — 2 digits

  Used to capture the bandwidth of the wideband calls to support H0, H11, H12, and N x 64 kbps data rates. For Enhanced Expanded, Enhanced Unformatted and customized record formats, this value is expressed as the number of DSOs of 64 Kbps channels comprising a call.

- **Bearer Capability Class** (bcc) — 1 digit

  This field contains the BCC for ISDN calls, identifying the type of an ISDN call. It will distinguish between voice and different types of data. The BCC is a single digit. Any one of the following may appear in this field.

  — 0 = Voice Grade Data and Voice

  — 1 = Mode 1 (56 kbps synchronous data)

  — 2 = Mode 2 (less than 19.2 kbps synchronous or asynchronous data)

  — 3 = Mode 3 (64 kbps data for LAPD protocol)

  — 4 = Mode 0 (64 kbps data clear)

  — w = Wideband

  Intraswitch CDR outputs a value in this field for Wideband calls only.

- **Calling Number** (calling-num) — up to 10 digits (15 for customized records)

  For outgoing or intraswitch calls, this field contains the extension number of the originating telephone user. For incoming and tandem calls, this field contains the TAC in standard formats. The fifth digit is the first digit of a 5-digit dialing plan. In formats where the field is less than 7 digits, this also shows the TAC of the incoming call.

This field shows the calling party number in Unformatted or Expanded records. If the CPN is not available, this field is blank for both formats.

This field contains the local extension of the NCA-TSC endpoint when the CDR record is for an outgoing (or originating) NCA-TSC. This field is blank for other NCA-TSC CDR records (that is, terminating, tandem, or unsuccessful).

- **Calling Number/Incoming TAC** (clg-num/in-tac)

  You can use this field on a customized record to display the calling number if it is available. If calling party number is not available, this field contains the Incoming TAC. For outgoing calls, this field contains the calling extension.

- **Carriage Return** (return)

  The ASCII carriage return character followed by a line feed indicates the end of a call record.

- **Condition Code** — 1 character

  The condition code indicates what type of call this record describes. For example, condition code C indicates a conference call, 7 indicates an ARS call, etc.

  Table 56 shows condition codes for most record formats. The 59-character format uses different condition codes from those used for other record types. The codes that apply to 59-character records appear in parentheses in the table.

Table 56. Condition Codes

Condition Codes	Description
0	Identifies an intraswitch call (a call that originates and terminates on the switch).
1 (A)	Identifies an attendant-handled call or an attendant-assisted call (except conference calls).
4 (D)	Identifies an extremely long call (10 hours or more) or an extremely high message count TSC (9999 messages or more). On a call exceeding 10 hours, a call record with this condition code and a duration entry of 9 hours, 59 minutes, and 1–9 tenths of a minute is produced after the first period. A similar call record with this condition code is produced after each succeeding 10-hour period. When the call does terminate, a final call record with a different condition code identifying the call type is produced.

Continued on next page

Table 56. Condition Codes (Continued)

Condition Codes	Description
6 (E)	Identifies calls that are not recorded because of resource exhaustion. A record with this condition code is generated for calls that are routed to the attendant or calls that require CDR to overwrite records. This also includes ISDN calls that did not complete at the far end, for which the Q.931 message included a cause value. It does not include ISDN calls that receive inband tones. The record includes the time and duration of the outage.
7 (G)	Identifies calls served by the AAR or ARS Selection feature.
8 (H)	Identifies calls which have been served on a delayed basis via the Ringback Queuing feature.
9 (I)	Identifies an incoming or tandem call, or an incoming or tandem NCA-TSC.
A	Identifies an outgoing call.
B	Identifies an adjunct-placed outgoing call.
C (L)	Identifies a conference call. For trunk CDR, a separate call record with this condition code is produced for each incoming or outgoing trunk serving the conference connection. The only extension recorded for a conference call is the conference call originator, provided ITCS and OTCS are disabled. For intraswitch CDR, if the originator is optioned for intraswitch, each time the originator dials a non-trunk party a separate call record is produced with this condition code, provided ITCS is disabled. If the originator is not optioned for intraswitch CDR, a separate record with this condition code is produced for each intraswitch party dialed.
E (N)	An ineffective call attempt due to facilities not being available, such as all trunks are busy and either no queuing exists or the queue is full on an outgoing call, or the called extension is busy or unassigned for an incoming call attempt. This also means an ISDN Call By Call Service Selection call was unsuccessful because of an administered trunk usage allocation plan. Incoming trunk calls to a busy terminal do *not* generate a CDR record.
F	Identifies an ineffective call attempt because of either insufficient calling privileges of the originator (assigned per FRL), ISDN calls rejected by the switch due to an NSF mismatch, or an authorization mismatch which prevents the completion of a data call.
G	Indicates a call terminating to a ringing station.

Continued on next page

Table 56. Condition Codes (Continued)

Condition Codes	Description
H	Indicates that a ringing call has been abandoned.
I	Indicates a call terminated to a busy station.
J	Indicates an incoming trunk call that is a new connection using ANF-PR (Additional Network Feature–Path Replacement, see QSIG) or DCS with Rerouting.
K	Indicates an outgoing trunk call that is a new connection using ANF-PR (Additional Network Feature–Path Replacement, see QSIG) or DCS with Rerouting.

If the trunk-group CDR Reports field is set to **ring**, CDR records the ring time to answer or abandon for incoming calls originated by the trunk group. In addition, CDR indicates if the incoming destination is busy. This record is separate from the normal call duration record printed for an answered call. This information is indicated by the condition code.

When an incoming call originated by a trunk group with this option set is terminated to an internal destination, the call is tracked from the time ringing feedback is given to the originator. If the call is answered, a CDR record is printed with the condition code “G” and the duration reflects the time between the start of ringing and the answer of the call. If the call is abandoned before being answered, the system prints a record with the condition code “H” and the duration reflects the time between the start of ringing and the time the call was abandoned. If the destination is busy, a CDR record is printed with the condition code “I” and a duration of 0.

- Condition Code overrides

  If two condition codes apply to the same call, one code overrides the other. The matrix below, Table 57 on page 1476, defines the overrides. To use this matrix, assume that condition codes 7 and A apply to the same call. To find the condition code that overrides, look at the point of where row 7 intersects column A (or where row A intersects column 7). In this case, condition code 7 overrides.

Table 57. Condition Code Override Matrix

CONDITION CODE

	0	**1**	**4**	**6**	**7**	**8**	**9**	**A**	**B**	**C**	**E**	**F**	**J**	**K**
0	NA	0	4	6	0	NA	NA	NA	B	C	NA	NA	NA	NA
1	0	NA	4	6	1	NA	9	1	B	C	E	NA	J	K
4	4	4	NA	6	4	4	4	4	4	4	NA	NA	J	K
6	6	6	6	NA	6	6	6	6	6	6	6	6	6	6
7	0	1	4	6	NA	7	9	7	B	C	E	F	J	K
8	NA	NA	4	6	7	NA	NA	8	B	C	E	NA	NA	K
9	NA	9	4	6	9	NA	NA	NA	NA	C	E	F	NA	NA
A	NA	1	4	6	7	8	NA	NA	B	C	E	F	NA	NA
B	B	B	4	6	B	B	NA	B	NA	B	E	F	NA	K
C	C	C	4	6	C	C	C	C	B	NA	NA	NA	J	K
E	NA	E	NA	6	E	E	E	E	E	NA	NA	NA	E	E
F	NA	NA	NA	6	F	NA	F	F	F	NA	NA	NA	F	F
J	NA	J	J	6	J	NA	NA	NA	NA	J	E	F	NA	NA
K	NA	K	K	6	K	K	NA	NA	NA	K	E	F	NA	NA

- **Date**

  You can include the date in customized records only. The format is based on the value of the CDR Date Format field on the CDR System Parameters screen.

- **Dialed Number** (dialed-num) — up to 23 digits

  This field contains the number dialed. If it is an outgoing call, the field contains the number dialed by a system user. If it is an incoming call, the field contains the extension that was dialed (or implied, as in Dialed Number Identification System). If more than 18 digits are dialed, the least significant digits (starting from the right) are truncated.

  If CDR Privacy is active for the calling number and this is an outgoing call, the trailing digits of the dialed number are blank in the call record. If more than 18 digits are dialed, the system truncates the dialed number to 18 digits, then blanks the administered number of digits.

  For an outgoing (or originating) NCA-TSC or tandem NCA-TSC, this field contains the dialed digits used to establish a route to a far-end switch. It contains the extension of the local extension used as the NCA-TSC endpoint when it is for a terminating NCA-TSC. For an unsuccessful NCA-TSC, this field is blank.

  The # sign (or E for some formats) may appear in this field in the following cases for both ARS and TAC calls.

  — When the user dials **#** at the end of digit dialing

  — If an outgoing call experiences an interdigit-timeout interaction with the ARS Analysis table

  — When a user dials a TAC for a Look Ahead Interflow (LAI). For example: A successful LAI to <TAC> 1001 where 1001 is the remote VDN extension will yield **1001E** or **1001#** in the Dialed Number field. The # or E is used by the vector processing software to indicate the end of dialing.

  You can eliminate the # or E as the last digit of the CDR record using the CDR System Parameters screen.

- **Duration** (duration or sec-dur) — 4 digits

  This is the duration of the call, recorded in hours (0–9), minutes (00–59), and to tenths of minutes (0–9). Calls are rounded down in 6-second increments. Therefore, a call of 5-second duration will be indicated as 0 duration. If 9999 appears in this field, this call was in progress when a time change was made in the switch.

  You can use the customized record format to have the duration reported in hours/minutes/seconds. This field is called sec_dur.

- **Feature Flag** (feat-flag) — 1 digit

  The feature flag indicates whether a call received network answer supervision, and if the call was interworked in the network. The call duration starts at the point of receiving the network answer.

  You can administer the feature flag (on the CDR System Parameters screen) to reflect whether an outgoing ISDN call was reported as interworked by the network.

  — A 0 in this field indicates a voice call without network answer supervision, or NCA-TSC not established.

  — A 1 in this field indicates a data call without network answer supervision.

  — A 2 in this field indicates a voice call with network answer supervision, but interworked.

  — A 3 in this field indicates a data call with network answer supervision, but interworked.

  — A 4 in this field indicates a voice call with network answer supervision.

  — A 5 in this field indicates a data call with network answer supervision.

  If the feature flag indicates that the call received network answer supervision, then the time of answer is accurate, and the recorded duration is also accurate. If a call does not receive network answer supervision, or receives answer supervision but is interworked with non-ISDN trunks, the time of answer is not necessarily accurate. Therefore the recorded duration for these calls may also not be entirely accurate.

  Calls are considered data calls if they use a conversion resource (such as a modem) and/or originate or terminate on a data module.

- **Format Code —** 2 digits

  This field contains 2 values: 00 is no PPM; 03 denotes a PPM count in the digits record.

- **FRL** — 1 digit

  FRLs, numbered 0–7, are associated with the AAR and ARS features and define calling privileges. The information contained in this field is as follows:

  — If the call is an outgoing call and an authorization code is not used to make the call, this field contains the originating telephone user’s FRL.

— If the call is an outgoing call and an authorization code is used to make the call, this field contains the FRL associated with the dialed authorization code.

— If the call is an incoming or tandem call, this field contains the FRL assigned to the incoming trunk group.

— If the call is an incoming tandem tie trunk call, this field contains either the FRL assigned to the tandem tie trunk or the raveling class mark (TCM) sent with the tandem tie trunk call, depending on which was used to complete the call. On ISDN calls, this field always contains the TCM, if it was received.

You can administer CDR so that disconnect information appears in this field in place of the FRL. If you do this, for trunk CDR, the following disconnect information appears:

Data	Meaning
0	Cannot determine who dropped first
1	Switch party dropped first
2	CO dropped first
3	Maintenance seized the trunk

For intraswitch CDR, the following call disconnect data appears in this field in place of the FRL data:

Data	Meaning
0	Cannot determine who dropped first
1	Calling number dropped first
2	Dialed number dropped first

- **Incoming Circuit Identification** (in-crt-id) — 3 digits

  This field contains the member number of a trunk within a trunk group used for an incoming call. For outgoing calls, this field is blank. Tandem calls contain both incoming and outgoing circuit id-numbers.

  The format of this field varies from record to record. For printer, Teleseer and 59-character formats, the numbers appear inverted on the record. For example, the circuit ID 123 appears as 231 (tens, units, hundreds). If you want to change this to appear in hundreds, tens, units format (123), use the Modified Circuit ID Display field on the CDR System Parameters screen.

- **Incoming TAC** (in-trk-code) — 4 digits

  This field contains the access code of the incoming trunk group.

- **INS** (3 digits)

  This field specifies the ISDN Network Service requested for a call. This field applies only to ISDN calls. Each Network Specific Facility has a corresponding INS value, shown in Table 58.

  This field also appears as ISDN NSV (Network Service Value).

Table 58. Network Specific Facility to INS Mapping

Network Specific Facility	INS Value
OUTWATS Band 0	33
OUTWATS Band 1–255	34–288
Network Operator	324
Presubscribed Common Carrier Operator	325
Software Defined Network (SDN)	352
MEGACOM 800	353
MEGACOM	354
INWATS	355
Maximum Banded WATS	356
ACCUNET Digital Service	357
AT&T Long Distance Service	358
International 800	359
Multiquest	367

- **ISDN CC**

  The call charge supplied by the ISDN advice of charge function (see ‘‘Receiving call-charge information’’ on page 532).

- **ISDN NSV**

  See INS.

- **IXC Code**

  — Non-ISDN Formats — 1 digit hexadecimal

    Interexchange Carrier (IXC) codes,1–F hexadecimal, indicate the carrier used on the call. This information is sent to the CDR output device in ASCII code as a hexadecimal representation (for example, ASCII “F” equals “15”).

Users must dial an IXC access number to access a specific common carrier for a call. In the US, this number is in the form 10XXX, 950 — 1XXX, or any 8–11 digit number. The IXC access numbers applicable at a given location are associated with an IXC code on the Inter-Exchange Carrier Codes screen.

When ARS is used, and a route pattern inserts one of the administered IXC codes, the report contains the associated IXC code. If no IXC access number is used, or the carrier is selected at the CO, the report contains a 0.

— ISDN formats — 3 or 4 digits

With an ISDN record format, this field is a 3 or 4-digit field that identifies the actual IXC used on an ISDN call. This information is determined from the route pattern administration. For AAR and ARS calls, the 3-digit IXC value is administered in the route pattern for all ISDN calls. If a user dials an IXC code with a 10XXX format as administered on the Inter-Exchange Carrier Codes screen, the CDR record contains only the last 3 digits (4 for Enhanced). If a user dials a 7-digit IXC code, this field contains a 0.

- **Line Feed** — 1 character

  The ASCII line feed character follows a carriage return to terminate CDR records.

- **MA-UUI** — 1 digit

  Message Associated User-to-User Signaling shows the number of ISDN messages containing user data sent on an outgoing call. Data in this field can range from 0 to 9.

- **Node Number** (node-num) — 2 digits

  This field identifies the DCS node number of a switch within a DCS arrangement. The number output is the same as the node number on the Dial Plan screen (the local id).

- **Null —** 1 character

  The NULL is used to terminate and divide CDR Records (usually in triplets) when needed by the receiving adjunct.

- **Outgoing Circuit Identification** (out-crt-id) — 3 digits

  For outgoing calls, this field contains the member number of the trunk within a trunk group used. This field is blank for incoming calls. Tandem calls include both incoming and outgoing circuit id numbers. For outgoing and tandem NCA-TSCs, this field contains the signaling group used to carry the NCA-TSC.

The format of this field varies from record to record. For printer, Teleseer and 59-character formats, and the ISDN and enhanced forms of those records, the numbers appear inverted on the record. For example, the circuit ID 123 appears as 231 (tens, units, hundreds). If you want to change this to appear in hundreds, tens, units format (123), use the Modified Circuit ID Display field on the CDR System Parameters screen.

- **Packet Count (**tsc_ct**)** — 4 digits

  For ISDN TSCs, this field contains the number of ISDN-PRI USER INFO messages sent, received, or (for tandem TSCs) passing through the switch.

- **PPM**

  Periodic Pulse Metering (PPM) contains pulse counts transmitted over the trunk line from the serving CO. These are used to determine call charges.

- **Resource Flag** (res_flag) — 1 digit

  Indicates whether the call was circuit switched or packet switched, whether a conversion resource was used, or if the call involved a MASI terminal or trunk.

  — 0 — circuit switched, no conversion device used

  — 1 — packet switched, no conversion device used

  — 2 — circuit switched, conversion device used

  — 3 — packet switched, conversion device used

  — 8 — MASI call

- **Sec-dur**

  For customized records only, this field allows you to set the duration field to display seconds instead of tenths of minutes.

- **Space** — up to 40 characters

  The ASCII space character separates other CDR fields or fills unused record locations.

- **TSC-Count** (tsc_ct)

  This is the customized name for Packet Count. See Packet Count.

- **TSC Flag** (tsc_flag) — 1 digit

  This field describes call records that pertain to temporary signalling connections. When not equal to 0, this field will indicate the status of the TSC. Table 59 on page 1483 presents the TSC Flag encoding.

Table 59. Encoding for TSC Flag

Encoding	Meaning
0	Circuit-switched call without TSC requests
1–3	Reserved
4	Call Associated TSC requested and accepted in response to SETUP, no congestion control (applicable to originating node). Call Associated TSC received and accepted via SETUP, no congestion control (applicable to terminating node).
5	Call Associated TSC received and accepted via SETUP, congestion control (applicable to terminating node).
6	Call Associated TSC requested, accepted after SETUP, no congestion control (applicable to originating node). Call Associated TSC received and accepted after SETUP, no congestion control (applicable to terminating node).
7	Call Associated TSC received and accepted after SETUP, congestion control (applicable to terminating node).
8	Call Associated TSC requested, rejected (rejection came from outside the local switch).
9	Call Associated TSC requested, rejected (rejection came from the local switch, that is, lack of resource).
A	Non Call Associated TSC received, accepted, no congestion control (applicable to terminating node). Non Call Associated TSC received, accepted, no congestion control (applicable to terminating node).
B	Non Call Associated TSC requested, accepted, congestion control (applicable to originating node). Non Call Associated TSC received, accepted, congestion control (applicable to terminating node).
C	Non Call Associated TSC requested, rejected (rejection came from outside the local switch).
D	Non Call Associated TSC requested, rejected (rejection came from the local switch, that is, lack of resource).
E	Reserved for future use.
F	Reserved for future use.

- **Time**

  This fields contains the time that the call ended, or the time that a user dropped from a multi-party call, if Call Splitting is active.

- **VDN** (vdn) — 5 digits

  This field is only available on customized records. The call record contains the VDN extension number. If VDN Return Destination is active, this field contains the first VDN the caller accessed.

Security

Monitor call detail records daily for unusual calling patterns, long calls, international calls, calls outside of normal business hours, and other indications of toll fraud. Call accounting systems are available that automatically monitor CDR output for fraudulent calling patterns.

Considerations

- Date and Time

  If a time of day is not administered in the system, Avaya MultiVantage does not generate CDR records. If the time is changed while a call is in progress, the actual duration for that call is not reflected in the CDR record. Instead, a special sequence of 9999 is recorded in the CDR record to indicate that the call was in progress during a time change.

- Dial plan

  If the dial plan supports 6/7 digit extensions, only the formats that already support Calling Numbers longer than 7 digits (expanded, Unformatted, customized, and international ISDN) support 6 or 7 digits. All other formats send only 5 digits and when a 6 or 7 digit extension is involved, the last 5 digits are sent.

Interactions

- Abbreviated Dialing

  When a user places a call using Abbreviated Dialing or a Facility Busy Indicator button, all outpulsed digits appear on the record.

- Answer Detection

  The Avaya MultiVantage provides Answer Detection using a Call Classifier circuit pack. This feature is an option for your system and requires an 8-port call classifier circuit pack. CDR starts recording call duration at the time the answer is detected by the circuit pack.

- Attendant Console

  If an attendant-assisted call involves an outgoing trunk, the primary extension of the user who requests attendant service is recorded as the calling number, even if the attendant dialed the outside number. Condition Code 1 indicates the call was assisted by the attendant.

  If the attendant allows through dialing, the primary extension of the user who dialed the number is recorded as the calling party. Condition Code 1 indicates that a trunk access code was extended by the attendant. Condition Code 7 indicates that a feature access code was extended by the attendant.

  If Incoming or Outgoing Attendant Call Record is enabled, the system produces a separate record for the attendant portion of incoming or outgoing calls that are transferred.

  On attendant-assisted calls that require an account code, the account code must be entered before the trunk access code.

  If the attendant is redirecting an incoming call to an extension, the attendant may dial an account code before dialing the extension number.

  It is not possible to option the attendant for intraswitch calls. Intraswitch records are produced for an intraswitch-optioned extension calling the attendant or for a call from the attendant to an intraswitch-optioned extension. In the case of an attendant-assisted call involving an intraswitch extension, the calling number recorded is the extension of the party who called the attendant, and the dialed number recorded is the extension that the attendant extended the call to. The record has a Condition Code 0.

- AUDIX

  For remote AUDIX over DCS, if station A on node 1 forwards its calls to AUDIX on node 2, each switch produces a call record. The record from node 1 contains A as the dialed number. The record from node 2 contains AUDIX as the dialed number.

  If the calling number is on a different switch within the DCS network, or the call comes in over ISDN, the actual calling number is recorded in the Calling Number field, and the TAC of the trunk bringing the call into the local switch is recorded in the Incoming Trunk Access Code field of 24-word records.

  If the forwarded call is an incoming call, then, as in all cases (other than vectoring) in which an incoming call is forwarded, transferred, or conferenced using an outgoing trunk, two separate CDR records are produced, one for incoming and one for outgoing trunk usage. The outgoing trunk usage record lists AUDIX as the Calling Number.

- AUDIX - Transfer out of

  If Incoming Trunk Call Splitting is enabled, and Transfer out of AUDIX is used, CDR generates two records. The first contains the AUDIX, the second contains the transferred-to party.

- Authorization Codes

  Authorization codes are recorded on CDR records provided account codes do not exceed 5 digits for non-ISDN and ISDN LSU formats, or 4 digits for Enhanced LSU formats. On the 59-character CDR International Processing and International Direct records, the authorization code is never recorded. When account codes are dialed, for the non-ISDN and the ISDN LSU formats, authorization codes are recorded on CDR printouts if the account code length does not exceed 6 digits. For Enhanced LSU, the account code length must not exceed 6 digits.

- Automatic Selection of DID Numbers

  Incoming calls, if recorded at all, are recorded for the DID extension number, not the room extension number.

- AAR and ARS

  CDR records contain the following information for Automatic Route Selection (ARS):

  — Fact that an ARS call was made

  — Calling extension number

  — FRL of the calling extension

  — Called number

  — TAC of trunk group used for the ARS call

  — Time of call completion

  — Call duration (how long the parties talked)

  — IXC code, if any

  If CDR is suppressed for the trunk group actually used on an ARS call, a CDR record is not generated; otherwise, Condition Code 7 applies. The ARS access code is recorded in the Access Code Dialed field and the trunk access code for the trunk group actually used is recorded in the Access Code Used field.

  If an AAR call is placed to a busy trunk group and CDR is suppressed for that trunk group, the user hears reorder tone and the CDR output shows an ineffective call attempt.

If an ARS call is an attendant-assisted call, the CDR record shows the call with a Condition Code of 7 (ARS call) instead of a Condition Code of 1 (attendant-assisted call). This occurs because CDR is not notified until after the trunk is seized and, in this case, the trunk is not seized until the user dials the number.

For FEAC, if a trunk group is accessed via ARS, the trunk group's COR is not used to determine if an account code needs to be entered.

- Automatic Callback

  When the Automatic Callback feature is used for an intraswitch call, no CDR record is generated for the first call attempt or the ringback. However, if the caller or extension being called is optioned for intraswitch CDR, a record of the actual call is output provided the call is answered and completed.

- Automatic Circuit Assurance

  ACA calls generate intraswitch CDR if the terminating extension is monitored by CDR. The originating extension for ACA calls cannot be administered for intraswitch monitoring.

- Automatic Wakeup

  No CDR intraswitch records are generated for wakeup calls.

- Bridged Call Appearance

  CDR does not record any information on the party who bridges onto a call. Instead, the number that was called appears in the Dialed Number field of the CDR record. The duration of the call is recorded when the last party drops off the call. This also applies for intraswitch calls.

  If the user originates a call over a bridged appearance, the call record contains the calling number of the bridged appearance extension and not the extension number of the original, calling station.

- Busy Verification of Terminals and Trunks

  An attendant or user is never required to enter an account code when making a busy verification.

- Call-by-Call Service Selection

  When a successful call is made on a Call-by-Call Service Selection trunk, the network specific facility used on the call is translated into an INS number and recorded in the INS field of the call record. If a Call-by-Call Service Selection call is unsuccessful because of an administered trunk usage allocation plan, the INS number is recorded in the INS field of the report with a condition code of "E."

- Call Coverage

  When an incoming or intraswitch call is answered by a covering extension, the extension number dialed by the originating party is recorded as the dialed number. If a call is covered to an off-net location, the dialed number is the number of the off-net location, the calling number is the number of the station that is covered to the remote location.

- Call Forwarding All Calls

  When a call is forwarded to another extension, the extension number dialed by the calling party is recorded as the dialed number. If a call is forwarded to an off-net location, the dialed number is the number of the off-net location, the calling number is the number of the station that is forwarded to the remote location.

  CDR generates one record for a forwarded intraswitch call. In this record, the dialed number is the same as the extension dialed by the originating party.

  For a trunk call to a station that is forwarded to a trunk, CDR generates two records. The first record shows an incoming trunk call to the station. The second record shows an outgoing trunk call from the station.

  For FEAC, calls cannot be forwarded to a destination where a user is required to enter an account code.

- Call Park

  When a user parks an incoming or intraswitch call, that user’s extension is recorded as the dialed number in the CDR record. Call duration in CDR reflects the entire time the incoming trunk is busy (incoming) or until the call ends (intraswitch).

- Call Pickup

  When an incoming or intraswitch call is answered by another user in the pickup group, the extension number dialed by the calling party is recorded as the dialed number.

- Call Vectoring

  The CDR System Parameters screen can be administered so that the VDN extension is used in place of the Hunt Group or Member extension. If administered to do so, this overrides the Calls to Hunt Group - Record option of CDR for incoming Call Vectoring calls.

  Outgoing vector calls generate ordinary outgoing CDR records with the originating extension as the calling number.

For incoming calls to a VDN, the duration of the call is recorded from the time answer supervision is returned.

— If answer supervision is returned by the vector (via an announcement, collect, disconnect, or wait with music command), and the call never goes to another extension, the VDN extension is recorded as the called number in the CDR record.

— If the call terminates to a hunt group, the VDN, hunt group, or agent extension is recorded as the called number as per the administration discussed above.

— If the call terminates to a trunk, CDR generates the following two records:

- An incoming record with the incoming TAC as the dialed number.
- An outgoing record with the incoming TAC as the calling number and the digits dialed through the vector step as the dialed number.

If “member extensions” is administered on the CDR System Parameters screen and the call successfully completes to a station via the “route-to” command, the call record shows an incoming call to that station.

Call Vectoring “route to” commands that are unsuccessful do not generate ineffective call attempt records.

If a vector interacts with an extension or group that has Call Forwarding All Calls active, normal Call Forwarding/CDR interactions apply.

Some calls may originally look like intraswitch calls, but result in trunk calls (for example, a call from a station administered for intraswitch CDR to a VDN, which ends up an outgoing call on an outgoing trunk). Such calls will not generate intraswitch CDR records; the CDR record will have a condition code A - outgoing.

- Call Waiting Termination

  Call duration timing starts when the user answers an incoming call.

- Centralized Attendant Services

  If a CAS attendant extends a call for a user, and CDR is not assigned to the RLT trunk group, the user’s extension is recorded as the originator of the call. If the RLT trunk group does have CDR administered, the RLT trunk is recorded. If a CAS attendant answers a call but does not extend the call, no CDR records are made.

- CO Trunks

  All incoming and outgoing calls on a CO trunk group are recorded, if CDR is assigned to the trunk group and CDR is administered to record incoming calls.

- Conference

  For the purpose of CDR, a call is considered a conference call if it contains at least one trunk that is eligible for CDR plus two or more parties, or if it contains at least one party optioned for intraswitch CDR. Condition Code C applies to each CDR record made for a conference call.

  For a conference call, a separate CDR record is produced for each outgoing/incoming trunk serving the conference call. If ITCS or OTCS is enabled, CDR produces a separate record for each internal party on the call as well.

  For the outgoing portion of a conference call involving multiple extensions, the extension that requested outside dial tone to bring an outside party into the conference is recorded as the calling party.

  For the outgoing/incoming portion of a conference call, the call duration in CDR reflects the entire time the trunk is on the conference call.

  A separate CDR record is produced for each trunk used in a trunk-to-trunk transfer. If ITCS is active, the incoming trunk record shows the duration of the entire call.

  If the originator of the conference call is optioned for intraswitch CDR, each time the originator dials a non-trunk party, a new CDR record is started. For example, Station A is optioned for intraswitch CDR and calls Station B. Station A conferences in Station C. Station A drops from the call. Station B or C drops from the call. Two CDR records are output with Condition Code C: one for the A to B call and one for the A to C call.

  If the originator is not optioned for intraswitch CDR, but one or more parties brought into the conference are, one record with Condition Code C is generated for each dialed intraswitch party. For example, Station A calls Station B, which is optioned for intraswitch CDR. Station A conferences Station C. Station A drops from the call. Station B or C drops from the call. One CDR record is output with condition code C for the A to B call.

  Intraswitch conference call CDR records are output when both the calling number (originator) and dialed number (terminator) of the call drop. The duration of the call will be from the time the terminator answers until both the originator and terminator drop from the call.

  If the attendant originates the conference, only the dialed numbers corresponding to intraswitch optioned extensions stimulate the creation of CDR records.

- DCS

  Station information is not passed throughout the DCS network for CDR purposes.

- DID trunks

  All incoming calls on the DID trunk group will be recorded if administered to record incoming CDR and if CDR is administered for this trunk group.

- Emergency Access to the Attendant

  CDR does not generate intraswitch records for Emergency Access calls.

- Expert Agent Selection

  A logical extension can be assigned to an agent who can log into a phone using that extension number. On the CDR System Parameters screen, you can chose to record the agent’s logical extension as the called number rather than the hunt-group extension or hunt-group-member extension.

- FX Trunks

  All calls made on an FX trunk group are recorded if administered to record CDR and if CDR is administered for this trunk group.

- Hotline Service

  The stored number used on an outgoing or intraswitch Hotline call is recorded by CDR the same as if it was manually dialed.

- Hunt Groups

  Either the hunt group extension number or individual hunt group member extension number is recorded as the called number. This is administrable on the CDR System Parameters screen.

- Intercept Treatment

  If an outgoing or tandem call is routed to Intercept Treatment, the number dialed by the calling party is recorded as the dialed number, and Condition Code F is recorded.

- Inter-PBX Attendant Calls

  If a user calls an Inter-PBX attendant and the trunk group used has CDR assigned, call records contain the following information:

  — Condition Code — A

  — Access Code Dialed — Blank

  — Access Code Used — Trunk access code of trunk used

  — Dialed Digits — Inter-PBX attendant access code

- ISDN

  When true answer supervision is received from the network, an indication is sent to the CDR device to this effect. If an ISDN call has been interworked, the call record shows this, and answer supervision may or may not be accurate. If you use unformatted or expanded record formats, the SID/ANI appears in the record, if sent.

- Last Number Dialed

  The CDR access code and account code dialed are stored as part of the Last Number Dialed. However, some digits may be lost due to the limit on the number of digits stored for this feature.

- Manual Originating Line Service

  If an attendant establishes an outgoing call for a user, designated as a Manual Originating Line, the CDR record for the call is the same as for any attendant-assisted outgoing call. The calling extension is recorded as the calling number, and Condition Code 1 applies.

- Multiple LDNs

  If incoming call information is recorded, the called number recorded for LDN calls is the extension number or trunk group access code to which the attendant completes the call. If the call terminates at the attendant console only, the dialed number is the attendant extension. The attendant extension number is administrable (the default is 0).

  LDNs cannot be administered for intraswitch CDR. However, a call from an intraswitch optioned extension to a LDN produces an intraswitch CDR.

- Night Service

  The extension number assigned to the attendants is recorded as the dialed number. The attendant extension number is administrable (the default is 0).

- Off-Premises Station

  CDR data is recorded if the extension is involved in an outgoing/incoming trunk call or it (or the other terminal involved in the call) is optioned for intraswitch CDR.

- PCOL trunks

  An outgoing PCOL call shows the dialed number in the `Dialed Number` field of the CDR record rather than a TAC. An outgoing PCOL call is recorded as a call from the originating extension number via the trunk group associated with the PCOL. On incoming PCOL calls the answering extension’s primary extension is recorded as the called number if incoming calls are recorded.

- Planned Interchange

  When a planned interchange occurs (either demand or scheduled), it is possible for the CDR records on calls ending within 10–20 seconds after the interchange to report as “invalid long duration calls” (duration of 9:59:9 and Condition Code other than 4). This is caused by deviations in the clocks between the two processors and the short duration of the calls. Consider these records invalid.

- Private Network Access

  Private Network Access calls are recorded if CDR is assigned for incoming or outgoing tie trunks.

- Remote Access

  Remote Access calls are recorded if Remote Access is provided on a per trunk group basis (incoming destination is the remote access number), and those trunks are administered for CDR. The call record gives no indication that this is a remote access call, other than the trunk group access code.

- Ringback Queuing

  Condition Code 8 is recorded for an outgoing call which is queued for a trunk before completion. The length of time the call is queued is not recorded.

  When an outgoing call is queued for a trunk and is unsuccessful (the queue times out or the calling party does not answer the callback) a CDR record is not generated for the call.

- Security Violation Notification

  SVN calls generate intraswitch CDR if the terminating extension is monitored. You cannot administer the originating extension for intraswitch monitoring.

- Service Observing

  No CDR records are generated for Service Observing calls. Tandem Tie-Trunk Switching

  The calling party on an incoming trunk can dial the CDR account code. The `Calling Number` field in CDR is the trunk access code for the incoming trunk group, the called number is the number dialed.

- Temporary Bridged Appearance

  A CDR record is not affected by any second or subsequent use bridging a call.

- Temporary Signaling Connections

  Cal-associated TSCs and TSC requests appear in the call record, provided the switch is administered to use ISDN layouts. Non-call-associated TSCs and TSC requests generate separate CDR records if the switch is administered to record them. In either case, the `TSC Flag` and `Packet Count` fields of the call record contain TSC data.

- Tie Trunk Access

  Tie-trunk calls are recorded if CDR is administered to record the trunk group and to record incoming calls.

- Transfer

  If a user *originates* a call on an outgoing trunk and then transfers the call to another extension, the originating extension is recorded as the calling party.

  If a user *receives* a call on an incoming trunk and then transfers the call to another extension, the extension that originally received the call is recorded as the dialed number.

  If a user *receives an intraswitch call* and then transfers it to another extension, the extension that originally received the call is recorded as the dialed number.

  If call splitting is active, when a user receives or originates a trunk call and then transfers the call to another extension, two records are generated.

  Intraswitch CDR records are generated for each call to or from an intraswitch optioned extension. For example, Station A, which is intraswitch optioned, calls Station B. Station A then transfers the call to Station C. When either Station B or C drops, two CDR records with Condition Code 0 are output: one for the A to B call, and the second for the A to C call.

  Intraswitch CDR transfer records are output when both the calling number (originator) and dialed number (terminator) drop from the call. The duration of the call is from the time the terminator answers until both the originator and terminator have dropped from the call.

  If ITCS is enabled and an incoming trunk call is transferred to a local extension that is optioned for Intraswitch CDR, the call produces an incoming trunk call record. It does not produce an Intraswitch record.

  Users cannot dial an account code when transferring a call to another extension, unless they have console permissions. However, a user transferring a call to a trunk can dial an account code before dialing the ARS or TAC.

- Trunk-to-Trunk Transfer

  Although they are not really conference calls, Trunk-to-Trunk Transfer connections are treated as such for CDR purposes. A separate CDR record is generated for each trunk in the connection.

  Unanswered Trunk Calls may or may not be recorded depending on administration. You can administer each trunk group so that unanswered calls are recorded if they remain unanswered for a specified period of time.

  If Incoming Trunk Call Splitting is active, a trunk-to-trunk transfer produces a record of the incoming call, and a record of the outgoing call. The outgoing call record shows the duration from the time the call was transferred until both parties drop. The incoming call record shows the duration from the time the station receives the incoming trunk call until both parties drop.

- Uniform Dial Plan

  If one user calls another user via a UDP extension number, and the trunk group used has CDR assigned, CDR records the following information:

  — Condition Code — 7

  — Access Code Dialed — blank

  — Access Code Used — trunk access code of trunk used

  — Dialed Digits — Uniform Dial Plan extension

- VDN Return Destination

  An incoming call does not generate a CDR record until the originator drops from the call. CDR creates a record when a call goes to the return destination VDN, the originator has not dropped, and vector processing — that is, the return destination VDN — routes the call to an outgoing trunk. CDR does not create a record if vector processing routes a call from the return destination VDN to an internal call. The call record shows only the first VDN that the caller accessed, no matter how many other extensions are involved in the call.

  If an incoming VDN call is routed to a station, CDR includes the station in the record. If an incoming VDN call is routed to an outgoing trunk, CDR includes the VDN in the record.

Call Forwarding

Call Forwarding allows users to redirect calls to designated destinations. The forwarded-to destination can be an internal extension, external (off-net) number, an attendant group, or a specific attendant.

Call Forwarding provides five functions:

- Call Forwarding-All Calls — Allows a user to redirect every incoming call to the forwarded-to destination.
- Call Forward Busy/Don’t Answer — Allows a user to redirect incoming calls to a forwarded-to destination only when the user is busy or when the call is not answered after an administrable interval. If the extension is busy, the call forwards immediately. If the extension is not busy, the incoming call rings the called extension, then forwards only if it remains unanswered longer than the administered interval.
- Call Forwarding Off Net— Allows a user to forward calls to an off-net destination.
- When the Coverage of Calls Redirected Off-Net field is activated, calls can be monitored for call progress tones, and if a call is not answered, it returns to the Avaya MultiVantage for call coverage processing in some circumstances.
- Call Forwarding Override — Allows the user at the forwarded-to extension to override Call Forwarding at the forwarded-from extension on a per-call basis so the user can initiate a call or transfer a call back to the forwarded-from extension.

Detailed description

You assign Call Forwarding All Calls and Call Forwarding Busy/Don’t Answer to extensions on a Class of Service basis. You assign Call Forwarding Override and Call Forwarding Off-Net on a system-wide basis. You can also restrict Call Forwarding Off-Net with the Class of Service.

Call Forwarding All Calls

Phone users and data-terminal users can activate or deactivate Call Forwarding All Calls for their own terminals with a feature-access code or Call Forward-All feature button. Virtual extension users cannot activate or deactivate Call Forwarding All Calls. An attendant or phone user with console permission can activate or deactivate the feature for another extension, virtual extensions, TEG, DDC, UCD group, or ACD split (but not vector-controlled splits).

Call Forwarding Busy/Don’t Answer

The feature is activated or deactivated with a feature-access code or Call Forward Busy/Don’t Answer feature button. An attendant or phone with console permission can also activate or deactivate the feature for another extension by using a feature-access code. Virtual extension users cannot activate or deactivate Call Forwarding Busy/Don’t Answer.

Call Forward Busy/Don't Answer cannot be activated for hunt groups, data extensions, or terminating extension groups (TEG). Calls to an attendant or EAS agent cannot be forwarded.

Call Forwarding Off Net

When a call is forwarded off net, the forwarded-to number can have up to 18 digits. When counting the 18-digit limit, count the digits in the Trunk Access Code or AAR/ARS feature access code. Do not count the “#” used to terminate a forwarded-to number if the “#” is used.

If the Coverage of Calls Redirected Off-Net field on the System Parameters Customer-Options screen and the Coverage of Calls Redirected Off-Net Enabled field on the System Parameters - Call Coverage/Call Forwarding screen are both set to **y**, the system allows calls forwarded off-net to be monitored for busy or no-answer conditions. The system may bring the call back for call-coverage processing if the principal’s coverage criteria are satisfied at the forwarded-to destination. However, if the principal does not have a coverage path, the system does not monitor the call. It is left at the off-net destination regardless of whether it is answered or busy.

Calls forwarding off net require an available outgoing trunk. Additionally, when the Coverage of Calls Redirected Off-Net field is **y**, a call-classifier port may be required. If there are no call-classifier ports available when needed, the call still routes off net, but it is not monitored for call progress tones and cannot be returned to the switch for further call coverage processing.

Whenever an incoming trunk call is redirected off-net, a timer is set that precludes any other incoming trunk call from redirecting off-net until the timer either expires or is cancelled. The rationale for this mechanism is to prevent calls that were redirected off-net from being re-routed back to the original principal from the off-net destination, effectively creating a round-robin loop that continuously seizes trunks until they are exhausted. The blocked call receives busy tone or redirects to coverage.

NOTE:
If you send calls off-net and use the Call Classifier - Detector or the Tone-Clock (with Call Classifier-Tone Detector) circuit pack (the international version) for call classification and do not use the American tone plan, use the System-Parameters Country-Options screen to define specific country tones. If you use the Call Classifier - Detector or the Tone-Clock (with Call Classifier-Tone Detector) circuit pack (the international version) and do not use the System-Parameters Country-Options screen, your system will download the American tone plan regardless of your geographical location.

Call Forwarding — Override

You can administer Call Forwarding Override on a system-wide basis to allow a forwarded-to party to override Call Forwarding when placing a call to the forwarded-from party. Call Forwarding Override is invoked automatically if the system-wide override option is set.

Call Forwarding Override cannot be used when calls forward to an external number. Call Forwarding All Calls from a data user or a hunt group cannot be overridden with Call Forwarding Override.

Warning users if their calls are redirected

You can warn analog phone users if they have features active that may redirect calls. For example, if the user has activated call forwarding, you can administer a setting to play a special dial tone when the user goes off-hook. See ‘‘Special Dial Tone’’ on page 791 for more information.

Security

Use the **list call-forwarding** command to get general information and to identify unauthorized Call Forwarding feature activation. The list shows the station name, station number, and forwarded-to destination number.

SECURITY ALERT:
Users who do not have permission to call out of the building may not do so with Call Forwarding.

Considerations

Calls can be forwarded only once. Consider the following scenario. Extension A forwards its calls to extension B. Extension B forwards its calls to extension C. Calls made to extension A:

- Ring at A, if possible
- Ring at B, if possible
- Redirect to extension A’s coverage path, if available and A’s coverage criteria are satisfied when applied at B
- Are not forwarded to extension C

There is no maximum number of calls that can be forwarded simultaneously.

You can administer a phone to receive a redirection notification signal when a call is forwarded.

The system restricts users from forwarding calls to a number that they are not allowed to call.

If **save translation** is run after call forwarding is activated for a phone, forwarding status and destination number are saved to tape.

Attendant

Calls to attendants cannot be forwarded. However, calls can be forwarded to the attendant *group*.

The attendant cannot have a Call Forwarding button.

Only the attendant or phone user with console permission can activate Call Forwarding All Calls for Terminating Extension Groups (TEG), Uniform Call Distribution (UCD) groups, Direct Department Calling (DDC), and data modules.

Number of rings provided to a forwarded call

The following table shows the field used to specify the number of rings provided a forwarded call before the call redirects to coverage.

Type of Forwarding	Forwarding All	Forwarding Busy/Don't Answer
local	coverage path	local
remote	*	*
CCRON	off-net	off-net
DCS (CCRON off)	coverage path	local
DCS (CCRON on)	off-net	off-net

- Coverage path —- On the Coverage Path screen, the Number of Rings field.
- Local — On the System-Parameters Coverage/Forwarding screen, the Local Subsequent Redirection/CFWD Don't Answer Interval field.
- Off-net — On the System-Parameters Coverage/Forwarding screen, the Offnet Subsequent Redirection/CFWD Don't Answer Interval field.
- * — No coverage treatment applied (rings until answered).

Example

If you forward all calls to a local site, the switch checks the Number of Rings field on the ''Coverage Path'' screen.

If you forward busy/don't answer calls to a local site, the switch checks the Local Subsequent Redirection/CFWD Don't Answer Interval field on the ''System Parameters Call Coverage / Call Forwarding'' screen.

Interactions

- Answer Detection

  This feature shares call-classifier resources with the Coverage of Calls Redirected Off-Net feature.
- Attendant Override of Diversion Features

  If an attendant uses redirection override to call a user who has Call Forwarding active, the call does not forward and remains at the user's phone.

- Automatic Callback and Ringback Queuing

  Automatic Callback cannot be activated toward a phone that has Call Forwarding active. If Automatic Callback was activated first, the callback call attempt is redirected to the forwarded-to party.

- Bridging

  Calls do not terminate to bridged appearances when Call Forward Busy/Don’t Answer is active.

  The switch blocks users from bridging onto a call that has routed off net while the call is undergoing call classification.

- Call Coverage

  If the principal’s (forwarding extension) redirection criteria are met at the designated (forwarded-to) destination, the forwarded call redirects to the principal’s coverage path; the designated destination gets a temporary bridged appearance (except when it is off net), which remains active after the call is answered so that the designated extension can bridge onto the call if desired. The temporary bridge appearance remains until the caller hangs up.

  If Cover All Calls is part of the coverage redirection criteria and if Call Forwarding or Call Forwarding Off-Net is active at a phone, incoming priority calls forward to the designated destination; all other calls redirect according to the Call Coverage path. Non-priority calls are not directed off net.

  When a covering user has activated Call Forwarding, a coverage redirected call does not forward to the designated extension number. Instead, the call is redirected to the next point in the principal’s coverage path, if available. If no other coverage point is available, the call remains at the principal’s phone.

- Call Detail Recording

  When a call is forwarded to an off-net number, the call is recorded in CDR records as a call from the forwarding station. When the Coverage of Calls Redirected Off-Net field is **y**, a CDR record is generated only after the call has actually been answered off net.

  If forced entry of account codes is required, calls cannot be forwarded to off-net destinations.

- Call Forwarding All Calls and Call Forward Busy/Don’t Answer

  Call Forwarding All Calls and Call Forward Busy/Don't Answer cannot be active for the same phone at the same time.

- Call Park

  Calls can be parked on a forwarded-from extension even though Call Forwarding is active for that extension. If a forwarded-to extension user parks a call that had been forwarded to that extension, the call is normally parked on the forwarded-to extension, not the forwarded-from extension.

- Call Pickup/Directed Call Pickup

  If the Temporary Bridged Appearance on Call Pickup field is set to **y** on the Feature-Related System Parameters screen, a Temporary Bridged Appearance is maintained if the forwarded-to destination is a member of the same call pickup group as that of the forwarded from station. If the Temporary Bridged Appearance on Call Pickup field is set to **n**, a Temporary Bridged Appearance is not maintained.

- Call Prompting

  This feature shares call-classifier resources with the Coverage of Calls Redirected Off-Net feature.

- Call Visor ASAI

  This feature shares call-classifier resources with the Coverage of Calls Redirected Off-Net feature.

- Conference

  The switch blocks users from conferencing another party onto a call which has routed off net while the call is undergoing call classification. If any party on a call that has forwarded off net is on hold (due to the initiation of a conference), the call routes off net without undergoing call classification. This occurs even when the Coverage of Calls Redirected Off-Net field is **y**.

- Expert Agent Selection

  Agents logged in with EAS enabled cannot activate or deactivate Call Forwarding. The physical extension where the EAS agent is logged in can be forwarded, but the EAS agent must first log out. Then, the phone can be forwarded.

- Hold

  If any party is on hold when a forwarded-to call routes off net, that call does not undergo call classification, even when the Coverage of Calls Redirected Off-Net field is **y**.

- Intercom—Automatic

  When a phone with Intercom—Automatic is call forwarded to another phone, the auto-icom feature also forwards. Intercom—Automatic is not forwarded if a call forwards off-net.

- Interflow

  The Interflow feature allows ACD calls to be redirected from one split to a split on another switch or to another external location. This is accomplished by forwarding calls that are directed to the split extension to an off-notify-net location via the Call Forwarding All Calls feature.

- Intraflow

  Call forwarding can be used to unconditionally redirect ACD calls from a split to another destination on the same switch.

- Leave Word Calling

  LWC cannot be activated toward a phone that has Call Forwarding activated. If LWC was activated before the called phone user activated Call Forwarding, the callback call attempt is redirected to the forwarded-to party.

- MFC Signaling

  This feature shares call classification resources with the Coverage of Calls Redirected Off-Net feature.

- Personal Central Office Line

  PCOL calls cannot be forwarded.

- QSIG

  If a call is forwarded over an ISDN-PRI trunk administered with supplementary service protocol “b” (QSIG), then additional call information may be displayed.

- Send All Calls

  If an extension has both Send All Calls and Call Forwarding All Calls activated, calls to that extension that can immediately be redirected to coverage are redirected. However, other calls, such as Priority Calls, are forwarded to the designated extension.

  Activation of Send All Calls at the forwarded-to extension does not affect calls forwarded to that extension.

- Temporary Bridged Appearance

  The system maintains a Temporary Bridged Appearance for on-net calls after the call is answered or until the caller hangs up. However, for calls forwarded off-net, the system cannot maintain a Temporary Bridged Appearance. Once the call is redirected to the principal’s coverage path, the trunk to the off-net, forwarded-to, number is released.

- Traffic Reports Removed

  The list measurement tone-receiver traffic reports provide port usage for this feature.

- Transfer

  The switch blocks a user from transferring a call which has routed off net to another party while the call is undergoing call classification. If any party on a call that has routed off net is on hold (due to the initiation of a transfer), the call routes off net without undergoing call classification. This occurs even when the Coverage of Calls Redirected Off-Net field is **y**.

Related topics

Refer to ‘‘System Parameters Call Coverage / Call Forwarding’’ on page 1104 for information on setting the number of times an extension rings before the system redirects the call or to determine whether the forwarded-to phone can override call forwarding to allow calls to the forwarded-from phone.

Refer to ‘‘Feature Access Code’’ on page 752 for information on forwarding calls to an administered number.

Refer to ‘‘Class of Service’’ on page 650 for information about assigning extensions a Class of Service (COS) that allows call forwarding.

Refer to ‘‘Station’’ on page 1066 for information about button assignments.

Refer to ‘‘Setting up call forwarding’’ on page 197 for instructions on administering various types of automatic call forwarding.

Call Park

Call Park allows users to put a call on hold and then retrieve the call from any other telephone within the system.

You can set a system-wide expiration interval for parked calls. If a call is not answered within the interval, the parked call redirects to an attendant or to the user who activated Call Park (the parking user). Calls redirect to the attendant if the default “Loudspeaker Paging” option is assigned and to the parking user if the Deluxe Paging and Call Park Timeout to Originator option is assigned.

If no attendant or night service extension is administered, and if Night Service — Trunk Answer from Any Station is not administered, the expiration interval is ignored and the call remains parked.

If two parties are connected on a parked call, a third party can also answer the call before the interval expires, creating a 3-way conference.

The attendant console group can have common, shared extensions used exclusively for Call Park. These extensions are not assigned to a telephone, but are stored in system translations and used to park a call. The extensions are particularly useful when one party is paged at the request of another party. The caller is parked on a common shared extension and the extension is announced. The status lamp associated with the extension identifies “call parked” or “no call parked” (instead of active or idle status).

Call Park allows telephone users to answer a call at one extension, but complete the call at another extension. Call Park also allows users to answer a call at any telephone after being paged by a telephone user or an attendant.

Considerations

- Only one call per extension can be parked at a time, even if the extension has multiple call appearances. Conference calls with up to five parties can be parked; the sixth position must remain open for the retrieving party.
- Calls cannot be parked on a group extension. If a group member places a call in Call Park, the call is parked on the member’s extension. Group members can belong to the following:
  - — A coverage answer group
  - — A DDC group
  - — A terminating extension group
  - — A UCD group
- If all appearances on a parked telephone are busy and no attendant or night-service extensions are configured when the call park timeout expires:
  - — A parked call is dropped if no coverage path is assigned
  - — A parked call is not dropped if a coverage path is assigned.

Interactions

- Abbreviated Dialing

  This button allows users to park calls or retrieve parked calls by pressing a button, instead of using the buttons and access codes separately.
- Automatic Wakeup

  Automatic Wakeup calls cannot be parked.

- Bridged Call Appearance

  If a user, active on a bridged call appearance, activates Call Park, the call is parked on the primary extension associated with the bridged call appearance.

- Call Vectoring

  A call cannot be parked on a VDN extension. Also, a call that is undergoing vector processing cannot be parked.

- Code Calling Access

  When a paging party dials the Code Calling Access code and the paged user’s extension, the paging party is automatically parked on the paged party’s extension.

- Common Shared Extensions

  If an attendant parks a call on a shared extension and tenant partitioning is not active, then when the call park timeout occurs, the call returns to the attendant group.

  If an attendant parks a call on a shared extension and tenant partitioning is active, then when call park timeout occurs, the call returns to the attendant who parked the call.

  The setting of the Deluxe Paging and Call Park Timeout to Originator field of the Feature-Related System-Parameters screen does not effect this behavior.

- Conference

  Conference calls can be parked.

- Data Privacy and Data Restriction

  These features are automatically deactivated when a call is parked.

- Drop

  If a digital-telephone user parks a call and then pushes the drop button, the call is unparked. If the parked call is from an internal digital-telephone user, pushing the drop button does not drop the call. The parking user must hang up to drop the call.

- Loudspeaker Paging Access

  Calls to paging zones cannot be parked.

- Music-on-Hold

  If a parked call involves only one party, the parked user hears music-on-hold. The parking user also hears music after first parking the call and hearing confirmation tone.

- Remote Access

  A Remote Access caller cannot park a call. However, the Code Calling Access feature, an answering attendant, or a telephone user can park an incoming Remote Access call.

Related topics

- Feature Access Code (FAC) screen
- Feature-Related System Parameters screen
- Station screen (multiappearance phones)
- Console-Parameters screen

Call Pickup

Call Pickup and Directed Call Pickup allow a telephone user to answer calls that alert at other extension numbers within the user's specified call pickup group. Directed Call Pickup allows telephone users to pick up any call on the Avaya MultiVantage system.

Call Pickup

Establish a call pickup group so that when one member of a group is away, other members can answer the absent member's calls. A call pickup group usually consists of users who are located in the same area or who have similar functions.

To pick up another user's call, a user goes off-hook and dials the Call Pickup access code or presses a Call Pickup button.

If a user's telephone has a Call Pickup button and status lamp, then:

- The status lamp lights steadily when Call Pickup is used.
- If Call Pickup Alerting is activated, members' status lamps flash when a call comes in to any extension in the call pickup group. Group members other than the called party, can answer using Call Pickup. The called party can answer on the ringing call or bridged appearance.

  NOTE:
  Call Pickup Alerting for a telephone takes effect only when the Call Pickup status lamp is not lit. If Call Pickup is used to answer a call, the status lamp lights steadily and does not flash if there are additional calls to the call pickup group.

  Both Call Pickup and Call Appearance status buttons flash at the called party's telephone.

If calls ring at 2 or more telephones in a call pickup group and a group member presses the Call Pickup button, a distribution algorithm determines which call is answered. Thus, all call pickup group members are treated equally. Specifically, when a Call Pickup button is pressed, the system searches the group extension numbers until reaching an extension with a call eligible for Call Pickup. The next time a Call Pickup button is pressed, the system searches from the *next* extension number.

For example, if extension A has 2 calls ringing and extension B has 1 call ringing, and one of extension A's calls is answered with Call Pickup, then extension B's call is answered the next time Call Pickup is used. After extension B's call is answered, a user can answer the second call to extension A.

When multiple calls ring on a telephone and a group member activates Call Pickup, the call with the lowest call-appearance number is answered. For example, if calls ring on the second and fourth call-appearance button on a telephone and a user at another telephone activates Call Pickup, the call on the second call-appearance button is answered.

Directed Call Pickup

Directed Call Pickup functions like Call Pickup, except for the following:

- A user can answer an alerting call at any telephone on the system — the alerting and answering telephones need not be members of the same call pickup group.
- You grant users permission to have their calls answered or to answer others' calls with Directed Call Pickup on a per-telephone basis on the Class of Restriction screen.

Considerations

- A telephone can be a member of only one call pickup group.
- When a call pickup group member is away from his or her telephone and receives a call, other call pickup group members' telephones do not ring. Therefore Call Pickup is only useful if either:
  - Call Pickup Alerting is enabled and call pickup group members have telephones with Call Pickup buttons and status lamps.
  - Call pickup group members are in close proximity and can hear each other's telephones ring.
- Exclusion is not supported for pickup calls.

Interactions

- Abbreviated Dialing

  A user can store:

  — The Directed Call Pickup FAC plus a telephone address in an Abbreviated Dial button

  — The Directed Call Pickup FAC. (The user then dials each extension.)

- Attendant

  Attendant can use Directed Call Pickup, but other users cannot use the feature to answer a call alerting at an attendant's telephone.

- Automatic Callback and Ringback Queuing

  Neither call pickup group members or Directed Call Pickup users can answer Callback calls.

- Bridged Call Appearance

  — If Call Pickup Alerting is activated and a bridged call appearance rings on a call pickup group member's telephone, other group members cannot pick up the call.

  — If Call Pickup Alerting is not activated and a telephone rings on a bridged call appearance, group members can pick up the call.

  — If Temporary Bridged Appearance on Call Pickup is enabled, a temporary-bridged appearance is maintained at the called telephone. This allows the called party to bridge onto the call after it has been picked up by another call pickup group member.

  — Directed Call Pickup cannot be used to pick up a call alerting at a bridged call appearance.

- Call Coverage

  You can use Directed Call Pickup to answer a redirected call alerting at a covering user's telephone if there is a call-coverage temporary bridged appearance.

- Call Detail Recording

  The extension number dialed by the caller is recorded as the dialed number in CDR.

- Call Forwarding

  If Temporary Bridged Appearance on Call Pickup is enabled, a temporary bridged appearance is maintained if the forwarded-to telephone belongs to the same call pickup group as the forwarded-from telephone. If Temporary Bridged Appearance on Call Pickup is not enabled, a temporary bridged appearance is not maintained.

- Call Pickup Alerting

  If a user who is a member of a ringing telephone's pickup group uses the Direct Call Pickup to answer a call and the call is the only call ringing for any member of the pickup group, the Call Pickup Alerting lamp goes dark when the user picks up the call.

  If a user who is not a member of a pickup group uses Direct Call Pickup to answer a call, then Call Pickup Alerting does not apply.

- Call Waiting Termination

  You cannot use Call Pickup to pick up a Call Waiting call.

- Conference

  If the Call Pickup Alerting field is enabled and a call is picked up and conferenced into a conference call, the Call Pickup status lamp flashes if additional calls are available for Call Pickup.

- Consult

  If the Temporary Bridged Appearance on Call Pickup field is not enabled, the consult call from the covering user appears as an idle-call appearance.

- Expert Agent Selection

  EAS agents can use Directed Call Pickup to pick up a call or have their calls picked up. The agent's COR overrides the COR of the telephone where the agent is logged in.

  If both the telephone's COR and the logged-in agent's COR allow Directed Call Pickup, the user picking up the call can use either the telephone's extension or the agent's loginID.

- Hold

  If the Temporary Bridged Appearance on Call Pickup field is not enabled and a user puts a call answered with Directed Call Pickup on hold, the called party cannot answer the call because a temporary bridged appearance is maintained. If the Temporary Bridged Appearance on Call Pickup field is enabled, then:

  — A call picked up and placed on hold at an extension remains on that extension, even if the called party answers the call.

  — If the Call Pickup Alerting field is enabled and a call is picked up and placed on hold, the Call Pickup status lamp flashes if additional calls are available for Call Pickup.

- Hot Line Service and Manual Originating Line Service

  Telephones assigned these features can be members of a call pickup group and have calls picked up, but they cannot answer calls for other pickup group members.

- Intercom — Automatic/Dial

  If Call Pickup on Intercom Calls is activated, you can use Call Pickup and Directed Call Pickup to pick up Automatic Intercom calls. If it is not activated, Automatic Intercom calls cannot be picked up and the calls are not included in the call-pickup-alerting-count.

- Internal Automatic Answer

  Internal calls to a telephone in a call pickup group are eligible for IAA. If the called extension in a call pickup group has IAA activated, the call is answered automatically. An extension that has IAA cannot automatically answer calls to other telephones in its call pickup group.

  IAA-eligible calls to an IAA extension cannot be answered with Call Pickup because they are automatically answered at the called telephone. Any non-IAA-eligible calls, such as external calls that ring the IAA-active telephone, can be answered by members of that telephone's call pickup group.

- Malicious Call Trace

  You cannot use Directed Call Pickup to pick up an alerting MCT call at the MCT-Controller telephone.

- Multimedia Call Handling

  Do not use Call Pickup or Directed Call Pickup with a Multimedia data endpoint. However, calls alerting at the voice component of a multimedia complex can be picked up with Call Pickup or Directed Call Pickup.

- Privacy — Manual Exclusion

  In the following case, the called party is not dropped when Privacy — Manual Exclusion is activated.

  A call is made to Station A and Station B picks it up using Call Pickup. Station A bridges onto the call by going off-hook on its call appearance. Station B activates Privacy — Manual Exclusion.

- Tenant Partitioning

  Directed Call Pickup follows existing Tenant Partitioning. The feature does not function across tenant partitions unless specifically administered to do so.

- Terminating Extension

  You cannot use Directed Call Pickup to pick up a call alerting at a TEG extension number.

- Transfer

  If the Call Pickup Alerting field is enabled and a call is picked up and transferred, the Call Pickup status lamp flashes if additional calls are available for Call Pickup.

Call Waiting Termination

Call Waiting Termination notifies a user with a single-line telephone who is active on one call that a second call is waiting. Single-line telephone users can place a call on hold to answer a waiting call. After answering the waiting call, they can return to the held call or toggle back and forth between the two calls. A single-line, telephone user can connect to only one call at a time.

Generally, the single-line telephone user hears one quick burst of tone when a call from another telephone user is waiting, 2 quick bursts of tone when an attendant-handled or an outside call is waiting, and 3 quick bursts of tone when a Priority Call is waiting.

NOTE:
Special ring tones are not supported over Direct Inward Dialing (DID) facilities.

A priority call can wait for the telephone to become idle even if Call Waiting Termination is not activated. However, an attendant-handled call receives busy tone unless the Attendant Call Waiting Indication field is set to **y**.

You assign Call Waiting Termination on a per-telephone basis. For a virtual extension, call waiting is assigned on the physical station.

Considerations

- An analog telephone user must place the active call on soft hold and dial the Answer Hold-Unhold feature access code to answer the waiting call.
- If an analog single-line telephone has Call Waiting enabled and has initiated a conference call, Call Waiting is denied. For example, caller A (on an analog telephone) is talking to caller B, then flashes and is talking to caller C, and then flashes to conference B and C. Then, if caller D attempts to call caller A, Call Wait is denied.

Interactions

Call Waiting is denied when the following features are activated at the single-line telephone:

- Another Call Waiting Call
- Automatic Callback (to or from the telephone)
- Data Privacy
- Data Restriction

A Call Waiting call cannot be picked up by a Call Pickup group member or by directed call pick-up.

Call-by-Call Service Selection

Call-by-Call Service Selection enables a single ISDN trunk group to carry calls to a variety of services. It does not require that each trunk group be dedicated to a specific service. It allows you to set up various voice and data services and features for a particular call.

Call-by-Call Service Selection provides the following benefits:

- Cost reduction — Since many services share the same trunks, the total number of trunks can be reduced.
- Improved service — Features and services are less likely to be blocked.
- Simplified Networking — Network engineering is simplified because analysis of trunking needs can be done based on total traffic instead of on a per-service basis.
- Timely response to changes — With UAPs, the network does not have to be consulted.
- Measurement of Call-by-Call Service Selection calls

Brief description

Call-by-Call Service Selection uses the same route patterns and route preferences that are used by Automatic Alternate Routing (AAR), Automatic Route Selection (ARS), and Generalize Route Selection (GRS). The service or facility used on an outgoing Call-by-Call Service Selection call is determined by information assigned in the AAR/ARS/GRS route patterns.

You can allow a variety of services to use a single trunk group. The system obtains trunking efficiency by distributing traffic over all the available trunks. Then you can assign services that are used on incoming and outgoing Call-by-Call Service Selection calls. The system provides traffic measurements for each individual service administered for an ISDN Call-by-Call Service Selection trunk group.

A Call-by-Call Service Selection example is shown below.

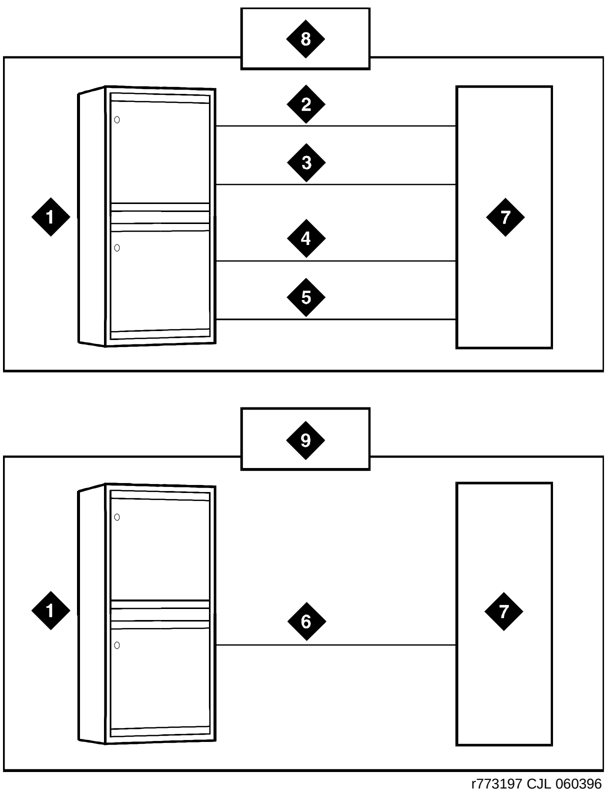


Figure Notes

1. Avaya Media Server
2. Megacom trunk group
3. Megacom 800 trunk group
4. SDN trunk group
5. OUTWATS trunk group
6. Call-by-Call Service Selection trunk group
7. Public-switched network
8. Without Call-by-Call Service Selection
9. With Call-by-Call Service Selection

Using Country Protocol 1, you integrate services and features such as MEGACOM, ACCUNET, and INWATS onto a single ISDN-PRI trunk group with flexible assignment of trunks to each service or feature. Calls such as an incoming 800 Service call that requires through-switching as an Outgoing WATS call can be routed via the same facility. See the ISDN trunk group screen on page ''ISDN trunk group'' on page 894 for a listing of available services.

NOTE:
When the Avaya Media Server is connected to 5ESS, all of the services supported under the Avaya and NI-2 country options are available.

Screens used to administer Call-by-Call Service Selection

You administer Call-by-Call Service Selection on a per trunk group basis. The following list shows the required screens and the fields you must use on each screen:

- System-Parameters Customer-Options
  - Version
  - ISDN-PRI
  - ISDN-BRI Trunks
  - Usage Allocation Enhancements
- ISDN trunk group
  - Service Type
  - Usage Alloc
  - all fields on the Incoming Call Handling Treatment (ICHT) Table
- Route Pattern
  - IXC
  - Service/Feature
  - Band

You can administer trunk Usage Allocation for multiple called numbers within a given Service/Feature, or you can administer trunk Usage Allocation for incoming or outgoing calls independent of Service/Feature.

- System-Parameters Customer-Options
  - Version
  - ISDN-PRI Trunks
  - ISDN-BRI Trunks
  - Usage Allocation Enhancements

- Network-Facilities (Refer to the DEFINTY services documentation for information about the Network Facilities screen.)
  - All

ISDN messages and information elements for usage allocation

Understanding the technical details of ISDN messages and information elements may help you implement ISDN.

Call-by-Call Service Selection allows the system to specify one of the preceding service types on a call-by-call basis. You can specify service types by classifying incoming calls to an ISDN Call-By-Call trunk group using the called-party's number.

You can also specify service types with a SETUP message that indicates the intent of the originating system to initiate a call using the specified service or facility. The SETUP message may contain units called information elements (IE) that specify call-related information. The IE used with Call-by-Call Service Selection are:

- Network-Specific Facility (NSF) — Indicates which facilities or services are to be used to complete the call (typically not used outside the US and Canada).

  The system also checks all incoming ISDN trunk calls for the presence of an NSF IE. If an NSF IE is present, the system makes sure that the requested service is compatible with the trunk administration before it accepts a call.

  For an outgoing call on a Call-By-Call trunk group, the NSF IE is constructed using the Service/Feature specified on the routing-pattern preference selected for the call.

  If the Service/Feature specified does not have an associated NSF, an NSF IE is not sent. For example, SETUP messages for incoming and outgoing calls classified only by a called-party number do not contain an NSF IE.

- Transit Network Selection — Indicates which interexchange carrier is to be used on an inter-LATA call.

  If a call requires both the Service/Feature and the interexchange carrier to be specified, the interexchange carrier information is sent in the NSF IE rather than the Transit Network Selection IE.

Usage allocation plans

Optional Usage Allocation Plans (UAP) may be assigned to provide more control over a Call-by-Call Service Selection trunk group. You can allocate a minimum and maximum number of channels for incoming and outgoing called numbers, privileged users, and voice and data calls.

A UAP allows the customer to set the following options:

- Maximum number of trunks that each service can use at any given time.The sum for all services may exceed the total number of trunk-group members. For example, for a15-member trunk group, you could administer a maximum of seven MEGACOM service calls, six MEGACOM 800 service calls, and eight SDN calls. This ensures that all trunk-group members are not dominated by a specific service, yet allows for fluctuations in demand.
- Minimum number of trunks that always must be available for each service. The sum for all services may not exceed the total number of trunk-group members. For example, for a 10-member trunk group that provides access to MEGACOM service, MEGACOM 800 service, and SDN, the minimum number of trunks to be used for each of these services cannot add up to more than 10.

When these UAP limits are exceeded, the system rejects the call, even if a trunk is available. On outgoing calls, the calling party receives a reorder tone unless other preferences are available.

You can assign either fixed or scheduled UAP for each Call-by-Call Service Selection trunk group.

- With a fixed UAP, one plan applies at all times.
- With a scheduled UAP, different plans can be administered to apply at different times of day and different days of the week. As many as 6 activation times and associated plans can be assigned for each day of the week.

You can have anything from a simple fixed UAP to a very flexible UAP with many scheduling options. You can even start out with no UAP and build one as the need arises.

Incoming call-handling treatment

Call-by-Call Service Selection provides special incoming call-handling treatment for ISDN trunk groups. An incoming call on an ISDN trunk is handled according to a treatment table administered for the trunk group. Depending on the platform you use, the table allows for a different number of combinations of call treatments.

The treatment for an incoming call is selected based on the first 3 columns in the ICHT table on the ISDN Trunk Group screen. When the attributes of an incoming call match these specifications, the call is treated according to the corresponding following 4 columns. If an incoming call matches more than one set of specifications, the most restrictive case applies. The following table lists the possible cases from most restrictive to least restrictive.

	Service / Feature	Called Len	Called Number
Most restrictive	Specified	Specified	x leading digits specified
	Specified	Specified	y leading digits specified, where y < x
	Specified	Specified	not specified
	Specified	Not specified	not specified
	“other”	Specified	x leading digits specified
	“other”	Specified	y leading digits specified, where y < x
	“other”	Specified	Not specified
Least restrictive	“other”	Not specified	Not specified

Call detail recording

On successful call attempts using ISDN Call-By-Call trunk groups, CDR records the NSF specified by the call’s NSF IE. CDR refers to this information as the ISDN Network Service (INS). The value passed to CDR is the 3-digit equivalent of NSF IE. NSF information for Facility Type 2 calls (used with ISDN-Pri Call-by-Call trunk groups) also is recorded if the NSF is available in the incoming SETUP message.

If an outgoing Call-by-Call Service Selection call uses an interexchange carrier other than the presubscribed common carrier, CDR records the 3-digit or 4-digit Interexchange Carrier Code (IXC). CDR may not record the IXC properly if the dialed-code format differs from the US IXC formats.

When a Call-by-Call Service Selection call is rejected because of a UAP, CDR records the cause as an ineffective call attempt. The NSF recording takes place also for the user-defined Facility Type 2. However, the NSF recording takes place only if the NSF is available in the incoming SETUP message.

Interactions

- Multiquest Flexible Billing

  Do not use a Service/Facility with the Facility Type field set to 2 or 3. NSF processing is not performed for Facility Type 2. An NSF is not included in the outgoing SETUP message for Facility Type 3.

- Time-of-Day Routing

  Any Time-of-Day Routing administration that affects routing preference also affects Call-by-Call Service Selection. Use Time-of-Day Routing to vary the IXC based on the time of day and day of the week.

Calling Party/Billing Number

Calling Party Number/Billing Number (CPN/BN) allows the system to request CPN/BN information from an AT&T network. The CPN is the calling party's telephone number. BN is the calling party's billing number. The CPN/BN may contain international country codes. CPN/BN can be used with an adjunct application.

Brief Description

The following list shows the screens used to administer CPN/BN and the fields you must use on each screen:

- ISDN trunk group
  - — All fields including Per Call CPN/BN
- ISDN Numbering — Public/ Unknown or ISDN Numbering — Private
  - — All
- Processor Channel Assignment (Refer to the DEFINTY services documentation for information about this screen.)
  - — All

On the ISDN Trunk Group screen, assign all fields needed to provide an ISDN-PRI link between the system and the adjunct. Set the Per Call CPN/BN field to **can-only**, **can-pref**, **bn-only**, or **bn-pref** as required to make sure that CPN/BN information is sent with a call-offered event report to the adjunct.

On the Processor Channel screen, assign all fields for one data link. The system can support only one interface.

Related topics

Refer to ''ISDN service'' on page 1610 for an overview of ISDN capabilities.

Refer to the *Avaya MultiVantage™ Call Center Software Release 11 Guide to ACD Call Centers* for information on using Calling Party Number and Billing Number with Automatic Call Distribution (ACD) and Inbound Call Management. Complete all screens required to administer ACD.

Refer to the DEFINTY services documentation for information about the Multifrequency-Signaling-Related System parameters screen.

Calling Party Number Restriction

With Calling Party Number (CPN) Restriction, you can administer individual phones to restrict sending CPN information on outgoing calls (per call restriction). Also, phone users can use a phone button or a feature access code (FAC) to restrict CPN information on individual outgoing calls (per line restriction). CPN Restriction works on any trunk that supports MFC signaling.

Per Line CPN Restriction overrides any outgoing trunk group CPN administration. Per Call CPN Restriction overrides any Per Line CPN Restriction for the phone, and it also overrides any Trunk Group administration for sending the calling number. For a tandemed ISDN call, only the Tandem Trunk Group's "sending calling number" administration applies.

Interactions

CPN Restriction is not supported on:

- local switch station-to-station calls
- calls originated from attendant console or data module
- DCS calls (DCS CPN information is not affected by CPN Restriction)
- Uniform Dial Plan (UDP) calls
- Trunk Access Code (TAC) calls where the Per Line CPN Restriction field on the * screen is **y** or **r**, or when the Per Call CPN Restriction FAC is dialed before the TAC
- non-ISDN calls that must be tandemed
- Per Call CPN Restriction over Adjunct Switch Application Interface (ASAI) and CTI interfaces

CLAN and processor CLAN QoS and CIDR support

Previous IP packet addressing and routing procedures for the TN799 Control-LAN circuit pack have not always been compatible with Avaya MultiVantage customers' evolved networks. Quality of Service (QoS) features are necessary to provide superior call quality for voice-over-IP (VoIP) signaling.

Although classful IP addressing structures were used in the beginning of VoIP development, enhancing C-LAN support of Classless Inter-Domain Routing (CIDR) and the closely-related Variable Length Subnet Mask (VLSM) paradigms provide the Avaya MultiVantage platforms with enhanced IP addressing and routing flexibility, while ensuring customer-network compatibility.

Configurations/Platforms

To use CIDR and IP QoS, the TN799DP Control-LAN circuit pack is required for:

- Avaya Media gateways (DEFINITY Server CSI, DEFINITY Server SI, DEFINITY Server R, S8700 Media Server, and S8700 Media Server)
- Avaya™ S8100 Media Server

To use CIDR and IP QoS, the TN799DP C-LAN circuit pack is not required for:

- Avaya S8300 Media Server
- MultiVantage™ S8100 Media Server with CMC1 Media Gateway
- MultiVantage™ S8100 Media Server with G600 Media Gateway

The S8300 Media Server uses the its processor CLAN feature. The MultiVantage™ S8100 Media Server with CMC1 Media Gateway and MultiVantage™ S8100 Media Server with G600 Media Gateway can use either its processor CLAN feature or a TN799DP Control-LAN circuit pack.

Windows Operating System

Whether the TN799 Control-LAN circuit pack tags frames with VLAN and user-priority values is administered on the IP Network Region screen. However, for the processor CLAN function on MultiVantage™ S8100 Media Server with CMC1 Media Gateway and MultiVantage™ S8100 Media Server with G600 Media Gateway, the operating system also controls tagging.

- If the IP Network Region screen is administered to tag, but the Windows operating system is set to not tag, then the processor CLAN function will not tag.
- If the Windows operating system is set to tag, but the IP Network Region screen is administered to not tag, then the MultiVantage™ S8100 Media Server with CMC1 Media Gateway or MultiVantage™ S8100 Media Server with G600 Media Gateway fill in the VLAN_ID and 802.1p fields of the extended ethernet header, but with the VLAN_ID to 0, and the priority to 0.

Tagging is administered on MultiVantage™ S8100 Media Server with CMC1 Media Gateway and MultiVantage™ S8100 Media Server with G600 Media Gateway as follows:

1. Log on to MultiVantage™ S8100 Media Server with G600 Media Gateway/MultiVantage™ S8100 Media Server with CMC1 Media Gateway using the Windows log-in.
2. Double-click the Intel PROSet icon in the Control Panel or in the system tray.
3. Select the first adapter from the list and right click on it.

   There are three adapters; PCLAN uses the first one.
4. In the menu that appears, click Add VLAN.

   You may see a dialog regarding 802.1 tagging. This tagging must be enabled for VLAN support.
5. Enter the VLAN ID and VLAN Name.

   The VLAN Name is for information only and does not have to match the name on the switch.
6. Click OK.

Linux Operating System

Whether the TN799 Control-LAN circuit pack tags frames with VLAN and user-priority values is administered on the IP Network Region screen. However, for the processor CLAN function on the S8300 Media Server with a G700 Media Gateway, the operating system also controls tagging. The IP Network Region screen and the bash command under the Linux operating system both set the same entries in a configuration file. There is no difference between setting them through one user interface versus the other.

Tagging is administered through the bash command line as follows:

vlanconfig -c -d device -v vlan_id [-i ip_address] [-g gw_address] [-m netmask] [-e on|off] [-f]

vlanconfig -r -d device -v vlan_id [-nf]

vlanconfig -q [-d device] [-v vlan_id]

-c	will create/change an interface
-r	will remove an interface
-q	will query an interface
-d device	create/change, remove, or query this device (valid with the -c, -r, and -q options)
-e on|off	enable or disable the interface (valid only with the -c option)
-f	force the command to execute (valid only with the -c and -r options)

Interactions

The IP Serviceability features do not interact with other Avaya MultiVantage switch elements.

- Distributed Communications System (DCS)

  Signaling to other switches or adjuncts can be done through the CLAN and the processor CLAN function.

  If you have an R11 connected to an older system that does not support 802.1p and 802.1Q VLANs on their signaling channels, you can administer it to be in a network region where these things are administered to “no.”

  If you decide to ignore the above advice and attempt signaling from a new system that is using 802.1p and 802.1Q VLANs on their signaling channels to an older system that does not support these protocols, then whether signaling information would:

  - Reach the destination, with the network using the frames’ VLAN and priority information, or
  - Reach the destination, but without having been given VLAN or priority treatment, or
  - Be dropped, either by the destination or by an intermediate device, depends on the capabilities of the switches or hubs used in the data network and how they are configured.

  The third possibility, “all frames dropped,” means no signals get through between the MultiVantage switch and the other device.

- ISDN QSIG

  ISDN QSIG signaling can be done through the CLAN and the processor CLAN function. If you have an R11 connected to an older system that does not support 802.1p and 802.1Q VLANs on their ISDN QSIG signaling channels, you can administer it to be in a network region where these things are administered to “no.”

  If you decide to ignore the above advice and attempt ISDN QSIG signaling from a new system that is using 802.1p and 802.1Q VLANs on their ISDN QSIG signaling channels to an older system that does not support these protocols, the CIDR and QoS feature will not work.

- Firmware Download of Circuit Packs

  File transfer for circuit pack firmware download is done through the CLAN. If you have an R11 connected to an older firmware FTP server that does not support DiffServ, 802.1p/Q, VLANs, and CIDR, you can administer it to be in a network region where these things are administered to “no.”

- H.323 Trunk

  H.323 IP trunk signaling is done through the CLAN and the processor CLAN function. If you have an R11 connected to an older system that does not support DiffServ, 802.1p/Q, VLANs, and CIDR on their H.323 trunk signaling channels, you can administer it to be in a network region where these things are administered to “no.”

Class of Restriction

You use Class of Restriction (COR) to define the types of calls your users can place and receive. Your system may have only a single COR, a COR with no restrictions, or as many CORs as necessary to effect the desired restrictions.

You will see the COR field in many different places throughout the MultiVantage System - when administering phones, trunks, agent logins, and data modules, to name a few. You must enter a COR on these screens, although you control the level of restriction the COR provides. You must administer a COR for the following objects:

- Agent LoginID
- Access Endpoint
- Announcements/Audio Sources
- Attendant Console
- Authorization Code — COR Mapping
- Console-Parameters
- Hunt Groups
- Loudspeaker Paging
- Data Modules
- Remote Access (each barrier code has a COR)
- Station
- Terminating Extension Group
- Trunk Groups
- Vector Directory Number

Called-party and calling-party restrictions

Called-party and calling-party restrictions are the basis for all CORs. When no restrictions are needed, assign a single COR with called-party and calling-party restrictions set to none. You can use this COR for unrestricted telephones, trunk groups, terminating extension groups, Uniform Call Distribution (UCD) groups, Direct Department Calling (DDC) groups, data modules, attendant groups, and individual attendant extensions.

The called-party restriction is checked only at the called terminal, module, attendant console, zone, or group, even if a call redirects from one telephone to another. For example, if a called terminal (with no terminal restrictions) has Call Forwarding active to a restricted terminal, the call still completes.

Inward restrictions

You can use inward restrictions to permit users to receive only internal calls. Inward restrictions prohibit users at assigned telephones from receiving public-network, attendant-originated, and attendant-extended calls.

The COR of the originally-called extension is the only one checked unless you administer 3-way COR check on conference and transfer calls. Denied calls are routed to intercept tone, a recorded announcement, or the attendant for Direct Inward Dialing (DID) calls.

Manual terminating line restrictions

You can use manual terminating line restrictions to allow users to receive calls only from an attendant or that were extended by an attendant. Calls can redirect to a manual terminating line-restricted telephone. The COR of the originally-called extension is the only one checked.

Local CO, foreign exchange (FX), and Wide Area Telecommunications Service (WATS) calls are routed to the attendant. DID calls are routed to an announcement or the attendant. Telephone calls are routed to intercept treatment.

Origination restrictions

You can use origination restrictions to prohibit users from originating calls. These users can still receive calls.

Outward restrictions

You can use outward restrictions to prevent users from placing calls to the public network. These users can still place calls to other telephone users, to the attendant, and over tie trunks. If necessary, an attendant or an unrestricted telephone user can extend a call to an outside number for an outward-restricted telephone user.

When outward restriction is applied to the Calling Party Restriction field on the Class of Restriction screen, calls coming into a trunk with that COR will be denied if they make use of the AAR/ARS feature.

Public restrictions

Public restrictions prohibit users from receiving public-network calls. Denied calls are routed to an intercept tone, a recorded announcement, or the attendant. Public restrictions still allow users to receive internal calls from other telephones or calls that were extended from the attendant.

Termination restrictions

You can use termination restrictions to prohibit users from receiving any calls. These users can still originate calls. DID or Advanced Private-Line Termination calls route to a recorded announcement or the attendant.

Fully restricted service

Fully restricted service prevents specific users from making or receiving public-network calls. Fully-restricted users cannot use authorization codes to deactivate this feature.

Calls from the public network to a fully-restricted extension redirect to intercept treatment or to the attendant. If the call redirects to the attendant, the attendant's display indicates the call was redirected because of fully restricted service (FULL).

There are circumstances where an extension with fully restricted service can access or be accessed by the public network.

Miscellaneous terminal restrictions

You can use miscellaneous terminal restrictions to prohibit users from accessing other specific terminals. Restricted calls are routed to intercept tone. Miscellaneous restriction groups apply on a per-COR basis. However, you can assign the same COR to more than one facility. Facilities with the same COR may be like facilities (such as two telephones) or different facilities (such as a telephone and a trunk group)

Miscellaneous trunk restrictions

You can use miscellaneous trunk restrictions to prohibit users from accessing specific trunk groups, such as WATS or CO trunk groups. Any or all trunk groups can be in a miscellaneous-trunk-restriction group. Restricted calls are routed to intercept tone.

Toll and TAC-Toll restrictions

Toll restrictions prevent users from placing public-network calls to certain toll-call numbers. Toll restriction is *not* a COR; you assign Toll restrictions to outgoing trunk groups on the Trunk Group screen. You disable TAC-toll restrictions for specific outgoing trunk groups on the Trunk Group screen.

Interactions

- AAR/ARS

  Originating FRLs are assigned via a COR. Termination and Miscellaneous Restrictions do not apply to ARS/AAR calls.

- AAR/ARS Partitioning

  Partition Group Numbers are assigned via a COR.

- Bridged Call Appearance

  The COR assigned to a telephone’s primary extension also applies to calls originated from a bridged call appearance of that extension on another terminal.

- Call Coverage

  Users who normally are restricted from calls can still receive calls directed to them via Call Coverage. When a call goes to coverage, the called party’s (not the covering party’s) restrictions are used.

- Call Forwarding All Calls

  If a call is restricted between the forwarding and forwarded-to extensions, Call Forwarding is denied. Restrictions are always checked when Call Forwarding is activated, but not when a call is forwarded.

- Call Vectoring

  When a call is directed to a VDN extension, the COR of the caller and the VDN are compared to determine if the caller can access the associated call vector.

- Class of Service

  In some cases, the COR can be overridden by the COS. See the Trk-to-Trk Restriction Override field documented with the ''Class of Service'' on page 650.

- Controlled Restriction

  Restrictions assigned via Controlled Restriction override COR restriction.

- Emergency Access to Attendant

  Emergency Access to Attendant calls are not restricted by COR.

Interactions for called-party and calling-party restrictions

- Night Service

  Night Service and Night Station — Trunk Answer From Any Station override Inward, Manual Terminating Line, and Public Restrictions.

- Tie-Trunk Access

  Incoming dial-repeating tie-trunk calls can be completed directly to an inward-restricted or public-restricted extension but cannot be extended by an attendant to an inward-restricted telephone.

- Transfer

  Incoming trunk calls cannot be transferred to an inward-restricted extension when a 3-way COR check is made.

  Incoming trunk calls can be transferred from an unrestricted extension to an inward-restricted or public-restricted extension if the 3-way COR check on Conference is overridden.

Interactions for fully restricted service restrictions

- Centralized Attendant Service

  Since COR information is not passed over Release Link Trunks (RLT), fully restricted service allows all CAS calls. Therefore, CAS allows a public network call to complete to a fully-restricted station.

- Distributed Communications System

  Fully Restricted Service allows all DCS calls because COR information is not transparent for DCS. DCS can allow a public network call to be completed to a Fully Restricted station.

- Power Failure Transfer

  All authorization features are bypassed when the switch is in Emergency Transfer Mode.

- Hunt Group

  The COR assigned to the Hunt Group is checked on calls redirected by the DDC or UCD of the hunt group. Extensions in the hunt group that have Fully Restricted Service can receive calls from the public network (via the hunt group) if the COR of the Hunt Group does not have Fully Restricted Service.

- Personal Central Office Line

  Do not assign fully restricted service to users who have a personal CO line. If you do, you will be paying for a CO line that no one can use!

- Remote Access

  If a barrier code is entered during connection to remote access, the code's associated COR is used for authorization checks. If remote access does not require a barrier code, then the default barrier code's COR is used. Remote Access can require an authorization code instead of or in addition to the barrier code. If an authorization code is required, the authorization code's associated COR overrides the barrier code's COR.

Do not assign fully restricted service to a station with the following features or conditions:

- Abbreviated Dialing
- Bridged Call Appearance
- Attendant stations
- Night Service stations
- Stations that are Call Coverage or Send All Calls points
- Stations that are Call Forward destinations
- Stations that are Call Pickup points

Interactions for miscellaneous terminal and trunk restrictions

- AAR/ARS

  AAR or ARS access to a trunk group overrides miscellaneous trunk restrictions.

- Abbreviated Dialing Privileged Group Number List

  A telephone user with authorization to access an Abbreviated Dialing Privileged Group Number List can place calls to any number on that list. COR assignments are not checked.

- Privileged System Number List

  A telephone user with authorization to access a Privileged System Number List can place calls to any number on that list. COR assignments are not checked.

Conference

The Conference button allows multiappearance telephone users to make up to six party conference calls without attendant assistance. This button also allows single-line telephone users to make up to three party conference calls without attendant assistance.

Considerations

- A single-line telephone can have up to 6 parties on a conference call, if each conferee adds another conferee. For example, one user can add a party, who then adds a third party, and so on.
- If you do not allow trunk-to-trunk connections and a telephone releases from a conference call (where all of the other parties were connected to the conference via trunks), then all parties are disconnected.
- If an analog single-line set has Call Wait active and creates a conference call, Call Wait is rendered inactive as long as the single-line set is on the call. For example, caller A on an analog set talks to caller B, flashes to talk to caller C, and flashes to conference B and C. Then, if caller D calls caller A, Call Wait is denied.
- Users of DCP, Hybrid, and wireless phones can conference a call on hold. If there is only one call on hold, no active call appearances, and an available call appearance for the conference, a user can initiate the conference process without taking the call out of hold. When the Conference button is pressed, Avaya MultiVantage assumes the conference is for the call on hold, and the conference feature works as usual.

  If there is more than one call on hold, the user must make a call active in order to include it in a conference. If the user presses the Conference button with two or more calls on hold, Avaya MultiVantage will ignore the conference attempt since it will not know which call the user wants to conference. If there are calls on hold and an active call, pressing the Conference button will start the conference process for the active call.
- You can allow users to abort a conference operation that is in progress by hanging up the phone. You set this parameter with the Abort Conference on Hang-Up field on the Feature-Related System Parameters screen. When a user hangs up the phone while trying to conference a call, the existing call is placed on hold, and the conference operation is aborted.

Conference/Transfer Toggle/Swap

Conference/Transfer Toggle/Swap allows users to toggle between two parties in the middle of setting up a conference prior to connecting all parties together or to consult with both parties prior to transferring a call. The display also will toggle between the two parties. This is not available on attendant consoles.

Selective Conference Party Display, Drop, and Mute

Selective Conference Party Display, Drop, and Mute allows any user on a digital station with display or on an attendant console to use the display to identify all of the other parties on a two-party or conference call. The user would press a feature button while on the call that puts the station or console into conference display mode. The user then can scroll through the display of each party currently on the call by repeatedly pressing the feature button. The display would show the party's number and name (when available).

The user could then do either of the following:

- The user can selectively drop the party currently shown on the display with a single button push. This can be useful during conference calls when adding a party that does not answer and the call goes to voice mail.
- The user can selectively mute the party currently shown on the display with a single button push. This puts the selected party in "listen-only" mode. This can be useful during conference calls when a party puts the conference call on hold and everyone on the call is forced to listen to music-on-hold. The user can mute that party so the conference call can continue without interruption. The muted party can then rejoin the call by pressing the # key on their telephone.

Meet-me Conference

Meet-me Conference allows a person to set up a dial-in conference of up to six parties. Meet-me Conference uses Call Vectoring to process the setup of the conference call. Meet-me Conference can be assigned optionally to require an access code. If an access code is assigned, and if the vector is programmed to expect an access code, each user dialing in to the conference call must enter the correct access code to be added to the call. The Meet-me Conference extension can be dialed by any internal or remote access users, and by external parties if the extension number is part of the customer's DID block.

No Hold Conference

No Hold Conference allows a user that is actively on a call to call another party and set up a conference call without interrupting the current call. The called party is added to the conference call as soon as they answer. Once the conference call is established, the call proceeds like a normal conference call. This feature can be used on a multiple-line digital telephones.

Interactions

- Bridged Call Appearance

  A bridged call appearance can be used to make conference calls. A bridged appearance can bridge onto a conference call only if there were never 6 parties on the conference.

- Call Vectoring

  A call to a VDN can be included as a party in a conference call only after vector processing terminates for that call (for example, after a successful route-to command).

- Class of Restriction

  If the Restriction Override field is set to **all**, the COR of the party being added is always checked against the COR of the party controlling the add-on, but the new party's COR is not checked against any other conferee's CORs.

- Trunk-to-Trunk Transfer

  When a multifunction telephone (BRI/Digital/Hybrid) dials sufficient digits to route a call, but could route differently if additional digits were dialed, the telephone does not recognize the Conference or Transfer buttons. The user must delay dialing for 3 seconds or dial # to indicate that the call can be routed based on the digits already dialed. The Conference or Transfer buttons are then recognized and the switch completes the call.

- VDN in a Coverage Path

  Calls in an established conference will not cover to a VDN.

  Once a call covers to a VDN, a conference cannot be established until the call is delivered to an extension and vector processing ends.

Related Topics

Refer to the ''Feature-Related System Parameters'' on page 766 screen for the following conference-related fields:

Public Network Trunks on Conference Call

Conference Parties With Public Network Trunks

Conference Parties Without Public Network Trunks

Conference Tone

Abort Conference Upon Hang-Up

Restriction — Controlled

Controlled Restrictions allow a telephone user with console permission to activate or deactivate specific restrictions.

Detailed description

Use Controlled Restriction to administer the following restrictions:

- Outward — The telephone cannot place calls to the public network.
- Total — The telephone cannot place or receive calls. (Allowed calls include calls to a remote-access extension, terminating-trunk transmission tests, and Emergency Access to Attendant calls.)

  Direct Inward Dialing (DID) calls are routed to the attendant or a recorded announcement. All other calls receive intercept tone.
- Termination — The telephone cannot receive any calls. Incoming calls are routed to the attendant, are redirected via Call Coverage, or receive intercept treatment.
- Station-to-Station — The telephone cannot place or receive station-to-station calls.
- Toll — The telephone cannot place toll calls but can place free local calls.

> **NOTE:**
> Toll Restriction may be substituted for either the outward or station-to-station restrictions. Administer this option on the ''Feature-Related System Parameters'' on page 766.

To activate Controlled Restriction:

1. Dial the group or extension feature access code.
2. Dial the number for the type of restriction desired:
   - — 1 for outward/toll
   - — 2 for total
   - — 3 for termination
   - — 4 for station-to-station/toll
3. Dial the extension (Attendant Control — Extension) or the Class of Restriction (COR) for a group of telephones (Attendant Control — COR).

Interactions

- Call Coverage

  Controlled Restrictions are not checked for covering users.

- Call Forwarding

  Controlled Restrictions for the forwarded-to extension are checked when Call Forwarding All Calls is active.

- Class of Restriction

  All telephones with the same COR are affected by a group restriction. When a call is placed, both individual and group restrictions are checked.

- Priority Call

  If a a station user or a Station-to-Station Restricted user activates priority calling before they dial another station, they receive intercept tone. They receive this tone whether you set Controlled Station to Station Restriction on the Feature-Related System Parameters screen to **y** or to **n**.

- Uniform Call Distribution

  Calls dialed through the UDP are not restricted by Outward Restriction.

Crisis Alert

Crisis Alert notifies designated extensions when an emergency call is made, and indicates the origin of the emergency call. This information allows the attendant or other user to direct emergency-service response to the caller.

When a user places an emergency call, the system notifies the designated extensions with audible and visual alerting. Audible alerting sounds like an ambulance siren. Visual alerting consists of flashing of the CRSS-ALERT button lamp and display of the caller name and extension.

When crisis alerting is active at the attendant console, the console is in position-busy mode so that no other incoming calls interfere with the emergency call. The console can still originate calls. The attendant must press the POSITION-BUSY button to unbusy the console and then the CRSS-ALERT button to deactivate audible and visual alerting.

Multiple emergency calls

If an emergency call is made while another crisis alert is still active, the call will be placed in queue. If you have administered the system so that all users must respond, then every user must respond to every call. The calls may not necessarily queue in the order in which they were made.

If you have administered the system so that only one user must respond, the first crisis alert remains active at the phone where it was acknowledged. Any subsequent calls are queued to the next available station in the order in which they were made.

Alerting a digital pager

Crisis Alert to a Digital Pager allows users to receive crisis alert messages on a pager. When a crisis alert call is originated in an emergency situation, a message of 7 to 22 digits is sent to the pager and displays a crisis alert code, an extension or room number, and a main number (if one is entered) so the pagee knows the location from which the emergency call originated. At the same time, an emergency call connects over a CAMA trunk.

To receive a crisis alert message, you need to administer at least one attendant or digital set with a CRSS-ALRT button. With the Alert Pager field set to **y**, any station with a CRSS-ALRT button and a pager receives the correct alert.

> **NOTE:**
> The crisis alert call uses 2-4 trunks; 1 trunk for the actual call and 1-3 trunks to notify the pager(s) depending on the number of administered pagers.

Information about the alert can be viewed on the history report printed at the journal printer and the emergency log.

Considerations

- Only one crss-alert button is allowed per attendant console or digital station.
- Consoles without a crss-alert button do not receive emergency notification.
- If a user attempts to make an emergency call, but all trunks are busy, this call will not generate an alert. If Outgoing Trunk Queuing is enabled for a trunk group, the call will queue, but will not generate an alert.

Interactions

- Centralized Attendant Service

  If CAS is enabled, the alert still goes to the local attendant.

- Phone Display

  When an emergency call is made and a crisis alert station with a 27-character display is notified, only 17 characters of the name field appear on the first display line, followed by the extension. The second line contains the last 3 characters of the name field, followed by the word "EMERGENCY." Characters 18 through 24 of the name field do not appear at all.

  We recommend that you consider the display for emergency notification when you complete the name field on the station screen. Put the most important identifying information at the beginning of the field.

- Tenant Partitioning

  If tenant partitioning is active, attendants only receive emergency notification from callers within their partition. If there is no attendant assigned to a partition from which an emergency call originates, the switch still sends a record of the call to the journal printer.

- Terminal Self Administration

  Those users who have the ability to administer their own phones do not have the ability to disable a crisis alert button.

Dial Plan

The dial plan provides information to the switch on what to do with dialed digits. A table defines the intended use of a code beginning with a specific first digit or pair of digits. These digits tell the system how many digits to collect before processing the full digit string. For example, a digit string beginning with 8 may tell the system to wait for 4 more digits because this is the first digit of a 5-digit internal extension.

All feature access codes, extensions and trunk access codes must be consistent with the dial plan.

Detailed description

The dial plan provides information to the switch on what to do with dialed digits. A table defines the intended use of a code beginning with a specific first digit or pair of digits. These digits tell the system how many digits to collect before processing the full digit string.For example, a digit string beginning with 8 may tell the system to wait for 4 more digits because this is the first digit of a 5-digit internal extension. For more information about the tables that define the dial plan, see:

- Dial Plan Analysis Table
- Dial Plan Parameters screen

You can also administer a Uniform Dial Plan (UDP) as part of the dial plan to be shared among a group of switches. For more information, see Uniform Dial Plan. So that calls route to the desired switch, a UDP requires the following information:

- A PBX code, which represents the first 1 to 5 digits of an extension and can range from 0 to 9xxxxxx with a maximum of 50,000 PBX codes on DEFINITY Server R or 20,000 PBX codes on DEFINITY Server SI/DEFINITY Server CSI.
- An RNX, which is associated with the PBX code and is used to select an AAR pattern for the call. This information is required for each PBX code. The 3-digit RNX can be an AAR location code or, for ENP calls, an ENP code.
- A PBX ID (1 to 63), which represents a specific switch (optional).
- Whether or not the PBX code is local to this system (optional).

Considerations

- You cannot assign prefixed extensions longer than five digits (including prefix) to intercom lists.
- A trunk access code (TAC) and an extension can share a first digit only if the extension is shorter than the TAC.
- Although extensions with the same first digit can have different lengths, data-channel extensions must have the maximum number of digits to avoid timeout problems for data calls that the switch automatically sets up, for example, the Call Detail Recording (CDR) link.
- An extension and a Feature Access Code (FAC) can share the same first digit only if the extension is longer as long as they are not used for Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) faxes. These extensions work only within the switch; they do not work as remote uniform dialing plan (UDP) extensions.
- When you design your dial plan or add new information to your dial plan, be careful if you assign the same first digit to more than one Feature Access Codes (FAC). Your system may need to distinguish between FACs with the same first digit by using the Short Interdigit Timer field on the Feature Related System Parameters screen.

Interactions

All dial-access features and services provided by the system require the dial plan.

- Attendant Display and Telephone (Voice Terminal) Display

  Prefixed extensions display without the prefix. The return call button causes the prefix to dial, even though it does not display.

- ISDN-BRI

  When an ISDN-BRI station dials sufficient digits to route a call, but the call could route differently if additional digits were dialed, the station does not recognize the Conference or Transfer buttons. The user must delay dialing for 3 seconds or dial # to indicate that the call can be routed based on the digits already dialed. The Conference or Transfer buttons then are recognized and the switch completes the operation.

- MF Signaling

  Flexible numbering is supported in countries using R2-MFC trunk signaling without Group II tones. Different-length extensions can exist as long as the extensions have different first digits.

- Property Management System (PMS)

  PMS products accept only extensions of 5 digits or shorter. Therefore, a customer using PMS cannot use 6/7-digit extensions.

- Uniform Dial Plan

  The following limitations apply to a distributed communications system (DCS) environment:

  — Extensions that differ in length from the UDP do not distribute to other switches.

  — You can use only a single length in your UDP.

- Single-Digit Dialing
  - A prefixed extension is still made up of a prefix and an extension of up to five digits.
  - Mixed station numbering extensions can have 1 - 7 digits.

Distinctive Ringing

Distinctive Ringing provides several ringing cycles to help telephone users and attendants distinguish between incoming call types. Administer Distinctive Audible Alerting on Feature-Related System Parameters for internal, external, priority, and attendant originated calls. If the phone is a single-line analog, you have to set this on the Station screen for each user.

Detailed description

You can administer system-wide distinctive-ringing cycles for the three basic call types. Most installations use 1-burst for internal calls, 2-burst for external calls, and 3-burst for priority calls. There are also non-administrable ringing signals for Automatic and Dial Intercom calls, Manual Signaling, and Redirect Notification.

Normally if an internal phone user transfers an external call, the call rings as internal. You can set a feature parameter (Update Transferred Ringing Pattern) to make the call ring as an external call.

Considerations

- If Distinctive Ringing is disabled, the system generates a 1-burst repetitive tone for all incoming calls. This is useful for equipment interfaced by analog lines, especially if you use off-premises station.
- A single distinctive ring cycle is used for each new incoming call to an off-hook telephone or headset. The system alerts a CALLMASTER terminal with a single ring cycle whenever either the headset or the handset is plugged into the headset jack.

Interactions

- Personalized Ringing

  The called party hears the user-selected ringing pattern for the distinctive ring cycles.

DS1 Trunk Service

Digital Signal Level 1 (DS1) trunk service uses bit-oriented signaling (BOS) and multiplexes 24 channels (T1 service) or 32 channels (E1 service) into a single data stream. DS1 can be used for voice or voice-grade data and for data-transmission protocols. T1 trunk service multiplexes 24 channels into a single 1.544-Mbps data stream. E1 trunk service multiplexes 32 channels into a single 2.048-Mbps stream. Both T1 and E1 provide a digital interface for trunk groups.

For information about how to administer DS1 with enhanced administration, refer to ‘‘Signaling modes’’ on page 1545.

Brief description

DS1 trunk service provides a digital interface for the following trunks.

- Voice-grade DS1 tie trunks
- Alternate voice/data (AVD) DS1 tie trunks
- Robbed-bit AVD (RBAVD) DS1 tie trunks
- Digital Multiplexed Interface (DMI) tie trunks
- ISDN-PRI trunks
- Central Office (CO) trunks
- Foreign Exchange (FX) trunks
- Remote-access trunks

- Wide Area Telecommunications Service (WATS) trunks
- Direct Inward Dialing (DID) trunks
- Main/Satellite tie trunks
- Tie trunks that link Common-Control Switching Arrangement (CCSA) or Enhanced Private Switched Communications Service (EPSCS) networks
- Release-link trunks for Centralized Attendant Service (CAS)
- Access trunks
- Off-premises stations (also known as station-side DS1)
- Access endpoints

DS1 also provides the following functions in public and private networks:

- Electronic tandem networks (ETN) or tandem tie-trunk networks (TTTN)
- Direct access to local exchange carriers

DS1 operational and signaling parameters

The table below provides the recommended combination of parameters for each DS1 interface.

DS1 Circuit Packs	# Trunk Members	Bit Rate Mbps	Companding	Signaling Mode	Trunk Type[1]
TN722	1-23	1.544	mulaw	common-chan	Tie, DMI-BOS, CO[2]
	1-24			robbed-bit	Tie
TN722B	1-23	1.544	mulaw	common-chan	Tie, DMI-BOS, CO[2]
	1-24			robbed-bit	CO/DID/Tie
TN767D, E[3]	1-23	1.544	mulaw	common-chan	Tie, DMI-BOS, C0[2]
	1-24			robbed-bit	CO/DID/Tie
	1-23 (24th is D-chan)			isdn-pri[4]	ISDN
	1-24			isdn-ext[4]	ISDN

Continued on next page

DS1 Circuit Packs	# Trunk Members	Bit Rate Mbps	Companding	Signaling Mode	Trunk Type[1]
TN464B[5]	1-30	2.048	alaw/mulaw[6]	CAS	CO/DID/Tie
	1-31		alaw	isdn-pri[4]	ISDN
	1-31			isdn-ext[4]	ISDN
	1-23	1.544	mulaw	common-chan	Tie, DMI-BOS, CO[2]
TN464C, D, E, F[3], and TN2464	1-24			robbed-bit	CO/DID/Tie
	1-23 (24th is D-chan)	1.544[4]	alaw/mulaw	isdn-pri[4]	ISDN
	1-31 (16th is D-chan)	2.048			ISDN
	1-24	1.544[4]	alaw/mulaw	isdn-ext[4]	ISDN
	1-31	2.048			
	1-30	2.048	alaw/mulaw	CAS	CO/DID/Tie
TN2242	1-30	2.048	alaw/mulaw	CAS	Tie
	1-30[7]	2.048	alaw/mulaw	isdn-pri	ISDN

1. CO is any of the following trunk types: CO, FX, WATS.
   Tie is any of the following trunk types: access, tie, tandem, RLT, APLT.
2. Common-channel DS1 circuit packs used in CO trunk groups must have a trunk type of auto.
3. Integrated CSU functionality is available only with the TN767D and TN464E or later-suffix DS1 circuit packs. Enhanced ICSU functionality is available only with TN767E, TN464F, and later-suffix DS1 circuit packs.
4. Mixed-mode signaling is allowed. This means that if the signaling mode is isdn-ext or isdn-pri, a port from that circuit pack may be used in any trunk group that allows robbed-bit signaling.
5. The TN464B’s companding is based upon the system companding that you administer.
6. ISDN-PRI calls are not guaranteed to work for the TN464B if the system’s companding is set to mu-law.
7. The administered D-channel on the DS1 screen for ISDN-PRI cannot be a trunk group member.

Signaling modes

Common-channel signaling (CCS) is an industry-standard technique where any one of a group of channels carries the signals for the other channels. Avaya uses the 24th channel of a group for signaling. This signaling technique differs from 24-channel signaling. When the system is configured for Facility-Associated Signaling, 24-channel signaling uses the 24th channel in a DS1 facility to carry signals. This technique also is called clear channel, out-of-band, or alternate voice data (AVD) signaling.

Channel Associated Signaling (CAS) is similar to common-channel signaling and is used only when the Bit Rate is 2.048 Mbps (the trunk is used with an E1 interface). Signaling is carried on the 16th channel.

Common-channel signaling and channel associated signaling provide a maximum transmission rate of 64 Kbps for bearer channels.

Robbed-bit signaling is a per-channel signaling technique for transmitting signaling bits on each channel in a DS1 facility. The least-significant bit in every 6th transmitted information frame is removed and replaced by a signaling bit. This technique is also called in-band signaling. The maximum transmission rate for each bearer channel with robbed-bit signaling is 56 Kbps.

ISDN-PRI signaling is carried on the 24th channel for a 1.544 Mbps connection and on the 16th channel for a 2.048 Mbps connection.

Public network signaling administration for ISDN-PRI Layer 3

The table below shows Avaya MultiVantage public network access connections for ISDN-PRI Layer 3.

Admin value	Country	Protocol supported	B-channel mtce msg
1 - a	United States, Canada	AT&T TR 41449/ 41459 (tested with AT&T network, Canadian network, and MCI network)	Service
1 - b	United States	Bellcore TR 1268; NIUF.302; ANSI T1.607	Restart
1 - c	United States	NORTEL DMS-250 BCS36/IEC01	Service
1-d	United States	Telecordia SR-4287	Service

Continued on next page

Admin value	Country	Protocol supported	B-channel mtce msg
2a	Australia	AUSTEL TS014.1; Telecom Australia TPH 1856 National ISDN protocol	Restart
2b	Australia	ETSI ISDN protocol	Restart
3	Japan	NTT INS-NET	Restart
4	Italy	ETS 300 102	Restart
5	Netherlands	ETS 300 102	Restart
6	Singapore	ETS 300 102	Restart
7	Mexico	ETS 300 102	Restart
8	Belgium	ETS 300 102	Restart
9	Saudi Arabia	ETS 300 102	Restart
10 - a	United Kingdom	ETS 300 102 (for connection to DASS II/DPNSS through external converter)	Restart
10 - b	United Kingdom, Ireland	ETS 300 102 (Mercury); British Telecom ISDN 30; Telecom Eireann SWD 109	none
11	Spain	Telefonica ISDN Specification	Restart
12 - a	France	VN4 (French National PRI)	None
12 - b	France	ETS 300 102 modified according to P10-20, called Euronumeris	None
13 - a	Germany	FTZ 1 TR 6 (German National PRI)	None
13 - b	Germany	ETS 300 102	Restart
14	Czech Republic, Slovakia	ETS 300 102	Restart
15	Russia (CIS)	ETS 300 102	Restart
16	Argentina	ETS 300 102	Restart
17	Greece	ETS 300 102	Restart
18	China	ETS 300 102	Restart
19	Hong Kong	ETS 300 102	Restart
20	Thailand	ETS 300 102	Restart
21	Macedonia	ETS 300 102	Restart
22	Poland	ETS 300 102	Restart

Continued on next page

Admin value	Country	Protocol supported	B-channel mtce msg
23	Brazil	ETS 300 102	Restart
24	Nordic	ETS 300 102	Restart
25	South Africa	ETS 300 102	Restart
ETSI - a	Europe, New Zealand, etc.	ETS 300 102	Restart
ETSI - b		ETS 300 102	None

EC500 Extension to Cellular

EC500 is an integrated mobility solution that offers users the freedom to work anywhere, anytime, using any type of cellular or wireless phone. With EC500, calls to an office number are extended to a cellular phone, allowing users to receive work-related calls wherever they are and whenever needed. Additionally, the cellular phone can be administered so that when an individual calls into the office, the individual’s name and office phone number appear in the caller ID display of the phone being called.

Additional information

For additional information:

1. Access the Avaya website at http://avaya.com
2. Click **Support**.
3. Click **Online Services**.
4. Click **Documentation**.
5. Click **Recent Documents**.
6. Scroll down to the Wireless section.
7. Select **EC500 Extension to Cellular**.

   You now can select from:

   - Avaya EC500 Extension to Cellular Release 3 Troubleshooting Guide, Issue 3, 210-100-102
   - Avaya EC500 Extension to Cellular Release 3 Installation and Administration Guide, Issue 1, 210-100-500
   - Avaya EC500 Extension to Cellular Release 3 User Guide, Issue 3, 210-100-700

Emergency Access to the Attendant

Emergency Access to the Attendant alerts an attendant if a telephone remains off-hook for more than the administered period of time. It also enables a user to place an emergency call to an attendant.

Detailed description

Emergency calls can be placed automatically by the system or dialed by system users. Such calls can receive priority handling by the attendant.

Place emergency calls to the attendant in the following ways:

- Automatically by the system

  Assign a telephone the Off-Hook Alert option via class of service (COS). If the terminal is left off-hook until intercept timeout, the (administrable) off-hook alerting timer starts. If the terminal is still off-hook when the timer expires, an emergency call is automatically placed to the attendant.

- Dial access by a system user

  A user can place an emergency call to the attendant by dialing the Emergency Access to the Attendant feature-access code.

When an emergency call is placed, one of the available attendants receives visual and audible notification of the call. If all attendants are busy, the call enters a queue for emergency calls. Calls can be administered to redirect to another extension if the queue is full.

An emergency call causes the following to occur:

1. The system selects the first available attendant to receive the call.
2. The attendant hears the emergency tone and sees the lamp associated with the Emergency button, if assigned, light. If the console does not have emergency-tone capability, the attendant hears normal ringing and sees the display flash.
3. The attendant display shows:
   - Calling-party identification
   - Calling-party extension
   - How many emergency calls remain in queue

An audit record is created for each emergency call. This record includes:

- Extension where the call originated
- The attendant or attendant group that answered the call
- Time of the call
- The following known call results:
  - — Call Completed — Call answered at attendant or listed directory number (LDN) night extension.
  - — Queue Full — Emergency-access queue is full; tries to redirect the call to an emergency-access redirection extension.
  - — No Attd — No active attendants are available to receive the call; tries to redirect the call to an emergency-access redirection extension.
  - — Redirected Answered — Call is answered by the emergency-access redirection extension.
  - — No Redirection Ext. — Could not redirect the call to the emergency-access redirection extension because none are administered.
  - — Attd Night Service — System is in night service. Will try to redirect the call to attendant night service.
  - — Failed — Caller drops the call before it can be answered. Call was either waiting in the attendant emergency queue, ringing at an attendant console, or ringing at the LDN night extension.
  - — Redirected Abandoned — Caller drops the call before it can be answered. Call had been redirected to the emergency-access redirection extension.

You can generate an Emergency Access Summary Report of the emergency audit records. Schedule the report for printing once a day at a designated printer. If the switch has a journal printer Emergency Access to the Attendant audit records print as the calls occur.

You can monitor emergency-access calls by displaying them at the administration terminal. The command for listing emergency call events is **list emergency.** You can use a from and to time option with the command. For example, if you enter the command **list emergency 8:00am 12:00pm**, the report shows emergency call events that occurred during that interval.

Considerations

- The emergency tone cannot be silenced except by answering the emergency call.
- The system should have at least one day and one night attendant (or night service station) for this feature to be useful at all times.

Interactions

- Centralized Attendant Service

  For a branch with CAS in effect, an emergency call reroutes to the branch attendant group. If the branch does not have an attendant or if the branch is not in CAS Backup Service, the call is denied.

  If the branch PBX is in CAS Backup Service, an emergency call routes to the backup position and is handled as any other non-emergency call.
- COR

  An emergency call to the Attendant overrides all restrictions on the COR.
- Individual Attendant Access

  An emergency call cannot be placed to an individual attendant.

  Emergency calls have priority over other calls to an individual attendant, only if they are assigned a higher priority on the Console Parameters screen.
- Inter-PBX Attendant Service

  For branches with Inter-PBX Attendant Service in effect, an emergency call reroutes to the local attendant group. If the branch does not have an attendant or if the attendant is not on duty, the call is denied.
- Night Service

  When Night Service is in effect, emergency calls route to the night destination. Such calls are included on the Emergency Audit Record, and the call is designated as Emergency Night in the audit trail.

  When an attendant is in night service, you must assign either a night station or a redirect extension. Otherwise emergency calls to the night attendant hear a busy tone.
- Off-Hook Alerting automatically places an emergency call to the attendant.
- Priority Queue

  You can change the priority of emergency calls to equal or lower than that of other types of calls.

- Remote Access

  An emergency call cannot be placed through Remote Access.
- Restriction — Controlled

  An emergency call overrides any controlled restriction.

Related screens

- Class of Service (Off-Hook Alert)
- Console Parameters
  - Queue Priorities
- Feature-Related System Parameters
  - Reserved Slots for Attendant Priority Queue
  - Time before Off-Hook Alert
  - Emergency Access Redirection Extension
- Hospitality
  - Extension of Journal/Schedule Printer
  - Time of Scheduled Emergency Access Summary Report
- Attendant Console - Feature Button Assignments
  - em-acc-att
- Feature Access Code (FAC)
  - Emergency Access To Attendant Access Code

Emergency Calls

Enhancement to E911 includes the ability to report the Emergency Location Extension as the Calling Party Number (CPN) by manually correlating the CPN with the phone location, even if the phone is moved.

Detailed description

If the IP Emergency Calls field on the Station screen is set to extension, the MultiVantage system uses the Emergency Location Extension as the E911 CPN instead of the IP phone’s extension.

The Enhanced 911 (E911) systems installed in many jurisdictions in the U.S. are equipped so that by dialing a service code (911 in the U.S.), a victim of a fire, accident, crime, or medical emergency may quickly access a public safety agency that will dispatch the appropriate response. The public E911 system maintains a database that stores location and background information to aid public safety agencies in responding quickly with the appropriate assistance. Information about the calling party can be triggered by the transmission of a Caller's Emergency Service Identification (CESID) number over Centralized Automatic Message Accounting (CAMA) trunks when the call originates from behind a PBX. 911 calls from a station on a PBX not equipped with CAMA trunks (or an adjunct computer system associated with CAMA trunks) will not provide the E911 system with the information required to identify the location of the person placing the call. Instead, the E911 system will only be able to identify the location of the trunk termination at the PBX.

This feature allows transmission of identifying information in the form of the CESID over CAMA trunks as to the extension number of a DID station associated with the calling party. The calling party may be at or near a station on a remote port network, or may be at a remote location served by an off-premises station.

Considerations

- Emergency response personnel might go to the wrong location when extensions and IP phones are moved without notifying the system administrator.
- The Automatic Number Identification (ANI) that is sent to the Central Office (CO) might not be the same extension as that of the phone used to dial emergency personnel. If the call is disconnected and emergency personnel call back they will call the ANI and may not be able to reach the person who placed the call. However, if the emergency location phone had a crisis alert button, the return call would be answered by someone who was notified of the extension making the emergency call. That person could forward the return call from the emergency personnel to the extension that made the emergency call. See "Crisis Alert".
- Emergency response personnel can be sent to the wrong location during the time between an IP phone registering after a move and the time that the system administrator updates the emergency location extension field.
- Enhancements to E911 only applies to emergency calls on CAMA and ISDN trunks.
- Several call appearances should be provided on the last phone in the Emergency Location Extension's coverage path and station hunting path.
- Emergency Location Extensions should not include voice mail, automated attendant or announcement extensions.

Emergency Transfer

Emergency Transfer provides service to and from the local telephone company central office (CO) during a power failure or when service is impaired. Emergency Transfer is also called Power Failure Transfer; the terms are synonymous.

Detailed description

Emergency Transfer allows analog telephones (500- or 2500-type) to access the local CO and to answer incoming calls during a power failure.

Each Avaya DEFINITY Server cabinet supports Emergency Transfer panels via the AUX connectors on the rear panel. The transfer is initiated when:

- a transfer panel or associated cabinet loses power.
- someone manually activates the Emergency Transfer switch on the associated maintenance circuit pack
- the software determines that service for that cabinet is severely impaired

You cannot activate any other system features during a complete system power failure.

Emergency Transfer panels are available in multiples of five telephones, which may be pulse-dialing or touch-tone phones. You must use pulse dialing if the CO accepts dial pulses only. Each telephone can be connected to a separate CO.

When your system is not in Emergency Transfer mode, transfer phones can be used as regular telephone.

Interactions

- Night Service

  If a power failure occurs when the system is in night service, the system automatically returns to night service when power returns.

Extended User Administration of Redirected Calls

Extended User Administration of Redirected Calls allows you to change your lead-coverage path or your call forwarding from any local (on-site) or remote (off-site) location.

SECURITY ALERT:

Invalid extensions and invalid station security codes are logged as security violations. The extension or incoming trunk from which the command sequence was dialed, the Feature Access Code (FAC), and the dialed command string appear on the Monitor Security-Violations Station Security Codes screen or report if Security Violation Notification is enabled.

For administration information about tracking security violations, refer to ''Setting up security violations notification'' on page 395.

Detailed description

This feature does not change Call Coverage, nor does it change Call Forwarding All Calls or the Call Forwarding Busy/Don't Answer. It merely allows you to select between one of two previously administered coverage paths or to change your forwarding from any on-site or off-site location.

Telecommuting access extension

The telecommuting access extension allows you to use Extended User Administration of Redirected Calls from off-site. If you want to quickly disable the feature for all users, change the Telecommuting Access Extension to blank.

If you are operating in a Distributed Communications System (DCS) environment, you need to assign a different telecommuting-access extension to each switch and tell your users which extension they should use. You can use Extended User Administration of Redirected Calls from any of the DCS nodes, but you need to dial the telecommuting-access extension of the node on which your station is defined before using the feature access code.

Extended User Administration of Redirected Calls and COS

The following table shows the relationship between class of service (COS) and a your ability to use call forwarding at the your station without a security code or from any on-site or off-site location with a security code.

Table 60. COS and Extended User Administration of Redirected Calls of Call Forwarding

If the user's COS are set to these values				Then the user's call forwarding capability is	
Call Fwd All COS	**Call Fwd B/DA COS**	**Extended Call Fwd Activate All COS**	**Extended Call Fwd Activate Busy D/A COS**	**Users can forward calls from their station without a security code**	**Users can forward calls from their station or from another location with a security code**
Yes	Yes	Yes	Yes	Yes	Yes
No	No	Yes	Yes	No	Yes
Yes	Yes	No	No	Yes	No

COR

Class of Restriction (COR) controls the use of the change coverage option of Extended User Administration of Redirected Calls. This means that, if the Can Change Coverage field on the COR screen is set to **y**, you can use the Change Coverage FAC to change your coverage option.

How to use from an attendant or console-permissions station

To use Extended User Administration of Redirected Calls from an attendant or console-permissions station is the same as that described for a user, except console-permissions stations and attendants do not need to enter a station security code, nor do they need to press the pound key (#).

How to use from an off-site location

To use Extended User Administration of Redirected Calls from off-site, you must first access the telecommuting-access extension. If you make the request via Direct Inward Dialing (DID), you must precede the extension with the correct public-network prefix. If you make the request via a trunk group dedicated to remote access of this feature, you must dial the public-network number for the trunk group.

The system provides dial tone after you access the telecommuting access extension. At that point you can enter only one of the four FACs associated with this feature. The 4 FACs associated with Extended User Administration of Redirected Calls are:

- Extended Call Fwd All Activate
- Extended Call Fwd Busy D/A Activate
- Extended Call Fwd Deactivation
- Change Coverage

When the system provides dial tone, you can proceed with the steps outlined for on-site use of the feature in ''Training users'' on page 367.

How to interrupt the command sequence

To interrupt the command sequence and begin again, you can enter an asterisk (*) at any point before the second pound sign. The system provides dial tone, and you can begin the command sequence at the point of entering your extension. (You should not enter the FAC again.) The interrupted command sequence is not recorded as an invalid attempt.

Interactions

- Bridged Appearance

  When the pound key (#) is pressed from a bridged appearance immediately following any of this feature's four FACs, the system assumes the currently active bridged extension will be administered. The station security code of the currently active bridged extension must be entered after the initial # to successfully complete the command sequence.

  If the station has only bridged appearances, the station's extension must be dialed after the FAC to successfully complete the command sequence, since the station's extension is not associated with any appearances.

- Call Coverage

  Extended User Administration of Redirected Calls interacts with this feature only in that system users can change their lead-coverage path.

  An attempt to activate Send All Calls is denied if the currently active coverage path does not allow Send All Calls in its coverage criteria. However, if you activate Send All Calls when it is allowed, and then change your coverage path to one that does not allow Send All Calls, the Send All Calls button remains lit and Send All Calls automatically resumes if you change back to the coverage path that allows it.

- Call Forwarding

  When Call Forwarding is active, the status lamps for the active features for that extension are lit. When Call Forwarding is deactivated, the status lamps for both Call Forward All and Call Forward Busy/DA buttons for that extension are extinguished. Off-net forward destinations are not allowed.

- Distributed Communications System

  Assign a different telecommuting access extension for each switch. You can use Extended User Administration of Redirected Calls from any of the DCS nodes, but you must dial the extension of the node on which your station is defined before dialing the FAC.

- Security Violation Notification

  Extended User Administration of Redirected Calls security violations are tracked and reported by SVN for station security codes, if it is enabled.

- Tenant Partitioning

  The telecommuting access extension is always automatically assigned to Tenant Partition 1, so it can be accessed by all tenants.

  The tenant number of the extension administered must be accessible by the tenant number from which the Extended User Administration of Redirected Calls FAC is dialed or the request is denied. If the FAC is being dialed on site, the tenant number of the station or attendant must have access to the tenant number of the extension administered. If the FAC is dialed off site, the tenant number of the incoming trunk must have access to the tenant number of the extension administered.

Related topics

Refer to ''Setting up telecommuting'' on page 351 for information about configuring telecommuting.

Refer to ''Telecommuting Access'' on page 1148 for information about and field descriptions on the telecommuting screen.

Refer to ''QSIG to DCS TSC Gateway screen'' on page 1029 for information about and field descriptions on the Remote Access screen.

Refer to ''Setting up remote access'' on page 364 for information about configuring remote access.

Refer to ''Feature Access Code'' on page 752 for information about and field descriptions on the Feature Access Code screen.

Refer to ''Class of Service'' on page 650 for information about and field descriptions on the Class of Service screen.

Refer to ''Station'' on page 1066 for information about and field descriptions on the Station screen.

Refer to ''Class of Restriction'' on page 636 for information about and field descriptions on the Class of Restriction screen.

Refer to ''Setting up call forwarding'' on page 358 for information about configuring call forwarding.

Refer to ''Assigning coverage options'' on page 357 for information about assigning two coverage options.

Facility and Non-Facility Associated Signaling

Facility Associated Signaling (FAS) allows an ISDN-PRI T1/E1 interface D-channel to carry signaling information for all the bearer (B) channels on its associated span.

Non-Facility Associated Signaling (NFAS) allows 1 ISDN-PRI T1/E1 interface D-channel to carry signaling information for up to 300 bearer (B) channels on its associated spans. In other words, a D-channel can carry signaling information for numerous B-channels located on different DS1 circuit packs.

NOTE:
NFAS is only valid for T1/E1 Country Protocol 1. Digital T1 service is also sometimes called "DS1" to distinguish it from analog T1 service.

ISDN-BRI trunks don't support Non-Facility Associated Signaling.

Brief Description

D-Channel Backup with NFAS

When NFAS is used, a backup D-channel can be administered to improve reliability. The system switches to the backup D-channel if a signaling link failure occurs on the primary D-channel span.

One D-channel is administered as the Primary D-channel and another D-channel is administered as the Secondary D-channel. These assignments ensure that both D-channels are in the same state at the same time and neither can be used to carry B-channel traffic at any time. The Primary D-channel is given precedence over the Secondary D-channel.

When D-Channel Backup is activated, all calls that have been answered are preserved. However, some call-related information may be lost. Calls that are not answered when D-channels are switched also may lose call-related information.

The figure below shows a possible configuration involving 3 ISDN-PRIs between an Avaya DEFINITY Server and another Avaya DEFINITY Server or the public network.

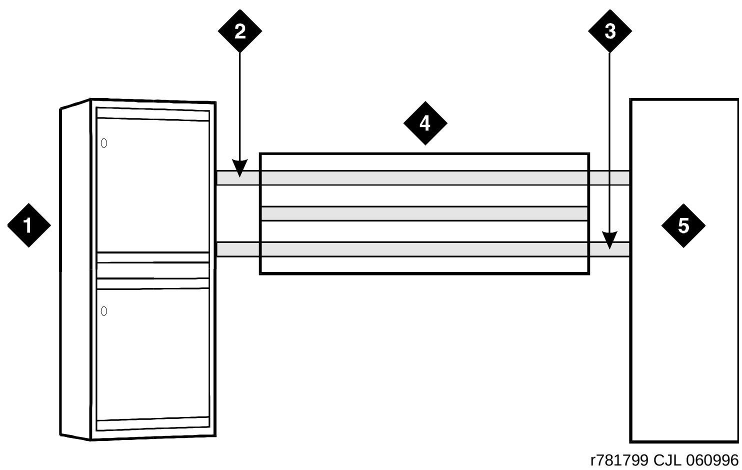


Figure Notes

1. Avaya Media Server
2. Secondary D-Channel
3. Primary D-Channel
4. ISDN-PRI controlled by D-Channel
5. Network Far-End Switch Avaya DEFINITY Server

Figure 58. ISDN-PRI Configuration

With T1 (24 channel) interfaces, 2 of the ISDN-PRIs contain a D-channel and 23 B-channels, while the other ISDN-PRI contains 24 B-channels. One of the D-channels is the Primary D-channel, and the other is the Secondary D-channel. Together, this pair of D-channels signals for all 70 (23+24+23) B-channels in the 3 Primary Rate Interfaces.

Since the D-channels are signaling for more than one ISDN-PRI facility, D-Channel Backup requires the use of NFAS. At any given time, one of the two D-channels is carrying Layer 3 signaling messages, while the other D-channel is active at layer 2, but in standby mode only. Any layer 3 messages received over the standby D-channel are ignored. Since only one of the D-channels can be active at a time, load sharing between the two D-channels is not possible. The two D-channels can provide signaling for only a predefined set of B-channels and cannot dynamically backup other D-channels on other interfaces.

D-Channel Backup activation

- D-Channel Failure

  If the signaling link fails on the active D-channel (D1) or the hardware carrying D1 fails, the system sends a message over the standby D-channel (D2). D2 then becomes the active D-channel and carries all subsequent signaling messages. When the signaling link or hardware on D1 recovers from the failure, D1 becomes the standby D-channel.

- System Technician Commands

  If a system technician commands that a D-channel switchover take place, the first action taken by the system is to tear down the signaling link on D1. After this is completed, a message is sent on D2 to request that D2 become the active D-channel. D2 then becomes the active D-channel and the switchover is complete.

Screens used to administer FAS and NFAS

The following list shows the required screens and the fields you must use on each screen:

- Signaling Group (Also refer to the DEFINTY services documentation for information about this screen.)
  — all
- ISDN trunk group
  — Port
  — Sig Grp
- DS1 Circuit Pack
  — Signaling Mode

Guidelines for administering FAS and NFAS

Coordinate the following when implementing FAS and NFAS:

- Decide which T1/E1 facilities will use FAS.
- Decide which of the remaining T1/E1 facilities carries D-channel signaling information on the 16th (E1) or 24th (T1) channel. For those channels that have a D-Channel Backup, D-channel pairs must be allocated.

- Define Signaling Groups. A Signaling Group is a group of B-channels for which a given D-channel (or D-channel pair) carries the signaling information. Each Signaling Group must be designated as either a FAS or NFAS Signaling Group.
  - A FAS Signaling Group must contain all the ISDN B-Channels on the T1/E1 interface associated with the group's D-channel, and cannot contain B-channels from any other DS1 circuit pack. For 24-channel DS1 boards, some of the DS1 ports may use in-band (robbed-bit) signaling and be members in a tie trunk group rather than an ISDN trunk group. These tie trunks cannot be members of a Signaling Group.
  - There is no restriction on which T1/E1 ports can belong to an NFAS Signaling Group. Normally, an NFAS Signaling Group consists of one or two D-channels and several complete T1/E1 interfaces.
  - If a Signaling Group contains only a subset of a T1/E1's B-channels (ports 1–12, for example), it is considered an NFAS Signaling Group, not a FAS Signaling Group. The remaining B-channels on the T1/E1 are then assigned as members of another NFAS Signaling Group.
- An Interface ID must be assigned to each T1/E1 facility in an NFAS Signaling Group. For example, if the B-channels in a Signaling Group span 3 T1/E1 facilities, a unique Interface ID must be assigned to each of the 3 facilities. This designation is required to uniquely identify the same B-channel (port) number on each of the T1/E1 facilities in the Signaling Group. Therefore, this interface must be agreed upon by both sides of the interface and administered prior to initialization.
- Primary and Secondary D-Channel Backup must be agreed upon by both sides of the interface and administered prior to initialization. If the IDs do not match, the signaling group will come up but calls will fail.

The following screens show the DS1 interface configuration for NFAS. When implementing FAS and NFAS, the DS1 screen must be submitted first, followed by the Interface Link and associated forms, followed by the ISDN-PRI trunk group, Signaling Group, and Trunk Group Members screens.

The Interface Link and associated screens may be administered at any time after the DS1 screens have been administered, with the following restrictions:

- A D-channel cannot be assigned on a Signaling Group screen unless the associated link is disabled.
- A trunk member cannot be assigned unless its associated Signaling Group has been administered.

The Signaling Mode field must be specified for each DS1 circuit pack. Because this circuit pack has the Signaling Mode field set to isdn-ext, all trunks on this circuit pack are signaled using either inband robbed-bit signaling or a D-channel on another DS1 circuit pack.

```
                    DS1 CIRCUIT PACK
           Location: 1B17                         Name: _______________
           Bit Rate: 2.048                            Line Coding: hdb3
     Signaling Mode: isdn-ext
Interface Companding: 5law
           Idle Code: 11111111
        MAINTENANCE PARAMETERS
      Slip Detection? n                     Remote Loop-Around Test? n
```

Screen 274. DS1 Circuit Pack screen

Next, Signaling Groups are administered using Signaling Group screens.

```
                              SIGNALING GROUP
Group Number : 1    Associated Signaling? n       Max number of NCA TSC: 0
                     Primary D-Channel: 1B1524     Max number of CA TSC: 0
                  Secondary D-Channel: 1B1624  Trunk Group for NCA TSC: __
        Trunk Group for Channel Selection: _____
    Trunk Brd    Interface ID          Trunk Brd    Interface ID
  1:   1B15          0             11:   ____            __
  2:   1B16          1             12:   ____            __
  3:   1B17          2             13:   ____            __
  4:   ____          __            14:   ____            __
  5:   ____          __            15:   ____            __
```

Screen 275. Signaling Group screen (Group 1) — D-channel Backup, Three DS1 Interfaces

```
                              SIGNALING GROUP
Group Number : 2   Associated Signaling? n       Max number of NCA TSC: 0
                    Primary D-Channel: 1B1824     Max number of CA TSC: 0
                 Secondary D-Channel: ______  Trunk Group for NCA TSC: __
       Trunk Group for Channel Selection: _____
    Trunk Brd    Interface ID          Trunk Brd    Interface ID
  1:   1B17          0             11:   ____            __
  2:   1B18          1             12:   ____            __
  3:   ____          __            13:   ____            __
  4:   ____          __            14:   ____            __
  5:   ____          __            15:   ____            __
```

Screen 276. Signaling Group screen (Group 2) — No D-channel Backup, Two DS1 Interfaces

```
                              SIGNALING GROUP
Group Number : 3    Associated Signaling? y      Max number of NCA TSC: 0
                       Primary D-Channel: 1B1924   Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC: __
        Trunk Group for Channel Selection: _____
```

Screen 277. Signaling Group screen (Group 3) — Facility Associated Signaling

Note the following details in the Signaling Group screens shown above:

- Signaling Group 1 B-channels on DS1 circuit packs (boards) B0 and B1 are signaled by D-channel pair B1524 (see the Primary D-channel field) and B1624 (see the Secondary D-channel field).
- Signaling Group 2 B-channels on board B1 are signaled by D-channel B1824.
- Board B0 has no D-channel. The B-channels on board B0 can be signaled by either D-channel pair B1524/B1624 (Signaling Group 1) or D-channel B1824 (Signaling Group 2).
- The DS1 interface on board B19 (Signaling Group 3) is a Facility Associated Signaling case. Note that the Secondary D-channel and Trunk Board/Interface ID fields are not displayed when the Associated Signaling field is **y**.

The following 2 communications-interface screens must be completed for the ISDN-PRI interface on DEFINITY Server SI configurations if the D-channel is switched through the TN765 Processor Interface (PI) circuit pack:

- Interface Links screen — Used to create an association between the D-channel on a DSI circuit pack and the port on a TN765 Processor Interface circuit pack used for this link.
- Processor Channels screen — Used to assign processor channels to the link administered on the Interface Links screen.

Finally, trunk ports are added to the ISDN-PRI trunk group and to Signaling Groups.

```
                                                         Page Y of X
                TRUNK GROUP
                         Administered Members (min/max): xxx/yyy
                               Total Administered Members: xxx
GROUP MEMBER ASSIGNMENTS
      Port              Code   Sfx     Name                   Night         Sig Grp
  1: 1B1501___                         ___________           _________        _
  2: 1B1523___                         ___________           _________        _
  3: 1B1601___                         ___________           _________        _
  4: 1B1623___                         ___________           _________        _
  5: 1B1701___                         ___________           _________        1
  6: 1B1709___                         ___________           _________        1
  7: 1B1716___                         ___________           _________        2
  8: 1B1724___                         ___________           _________        2
  9: 1B1801___                         ___________           _________        _
 10: 1B1823___                         ___________           _________        _
 11: 1B1901___                         ___________           _________        _
 12: 1B1923___                         ___________           _________        _
 13: _________                         ___________           _________        _
 14: _________                         ___________           _________        _
 15: _________                         ___________           _________        _
```

Screen 278. ISDN-PRI Trunk Group screen — Trunk Members with Required Signaling Group

The Sig Grp column on the above trunk group screen is completed as follows:

- If a DS1 interface appears in one and only one Signaling Group, then Sig Grp may be left blank because the system automatically populates the field with the correct Signaling Group.
- If a DS1 circuit pack appears in more than one Signaling Group, then the Signaling Group numbers must be entered in the appropriate fields before submitting the screen.

Related topics

Refer to ''ISDN service'' on page 1610 for an overview of ISDN capabilities.

Facility restriction levels and traveling class marks

Facility Restriction Levels (FRL) and Travelling Class Marks (TCM) allow certain calls to specific users, and deny the same calls to other users. For example, you can give certain users access to central office (CO) trunks to other corporate locations, and you can restrict other users to less expensive, private-network lines.

Detailed Description

FRL

The switch compares the FRL of the outgoing phone to the FRL of either the terminating trunk group or, for AAR and ARS, the routing preference specified on the Routing Pattern Table. If the FRL of the originator is equal to or greater than the terminating or route pattern FRL, the call continues. Otherwise, the call is blocked.

TCM

If an intertandem tie-trunk group is used for a call, then a TCM is outpulsed as the last digit. If the intertandem tie-trunk FRL is equal to or greater than the terminating FRL, the call continues. If the originating FRL is less than the terminating FRL, the TCM is compared with the tie-trunk's FRL. If the TCM is greater than or equal to the FRL, the call continues.

Call-originating facilities

Any of the following can originate an AAR or ARS call. Each is assigned an FRL via an associated Class of Restriction (COR).

- attendant
- data terminal capable of keyboard dialing
- incoming tie-trunk group from a subtending location
- incoming intertandem tie-trunk group (at a tandem switch)
- incoming access tie-trunk group (links a remote main switch to a tandem switch)
- phone
- remote access user

Phones and all incoming tie-trunk groups use the FRL of their COR. On attendant-extended calls, the attendant-group FRL is used. If Individual Attendant Access assigned, the individual attendant's COR FRL is used. Data terminals use the FRL of the COR assigned to the associated data module.

A remote access call uses the FRL of the COR assigned to the dialed barrier code. If a barrier code is not required on remote access calls, there is no FRL

Call terminating facilities

Any of the following trunk types can serve as the termination point for an AAR or ARS call:

- Tie trunk — excluding Release Link Trunks (RLT), but including Common Control Switching Arrangement (CCSA) and Enhanced Private Switched Communications Services (EPSCS) access trunks
- Wide Area Telecommunications Service (WATS)
- Central Office (CO)
- Foreign exchange (FX)
- Integrated Services Digital Network - Primary Rate Interface (ISDN-PRI)

Each of these outgoing trunk groups has an assigned COR that contains an FRL. However, this FRL is never used in an AAR or ARS call. A terminating-side FRL for AAR/ARS calls is assigned in the route pattern, not to the outgoing trunk group.

FRL guidelines

You assign the FRL to the trunk group within the route pattern. You can use the same trunk group in more than one route pattern, and the same trunk group can have a different FRL in a different pattern. You can assign the same FRL to more than one trunk group.

Be consistent in FRL assignments. For ease of assignments, always use FRL 0 or 1 for a trunk group that everyone can access. If you use a range of 0–5 in one pattern, use the same range in another pattern if all users can access the first-choice route.

Assign a COR with an FRL of 0 to a group of users to restrict them from making outgoing calls. Use any other number for the FRL on your first choice route pattern. This denies access to any trunk group for the users, because all trunk-group FRLs are greater than 0.

You assign FRLs for remote access users through the remote-access barrier codes. You can assign up to 10 barrier codes, each with its own COR and FRL. The simplest way to assign these FRLs is to duplicate the on-premises FRLs, then relate the appropriate barrier code to users who need remote access.

Example

The following is an example of how FRLs can be assigned in a COR:

- FRL0 — 911 access only
- FRL1 — Local calls only
- FRL2 — FRL1 plus home area-code calls using WATS
- FRL3 — FRL2 plus use of local lines for all calls in the home area code
- FRL4 — FRL3 plus calls to all the USA, using WATS only
- FRL5 — FRL4 plus calls to all the USA, using local lines
- FRL6 — FRL5 plus international calls
- FRL7 — Reserved

Interactions

- Call Detail Recording

  If 15-digit CDR account codes are used, the FRL field in the CDR record is overwritten with the account code.

Related topics

Refer to ''CLAN and processor CLAN QoS and CIDR support'' on page 1521 for more information the types of restrictions you can assign.

Refer to ''Route Pattern'' on page 1039 to find more information on fields on the route pattern screen.

Generalized route selection

Generalized Route Selection (GRS) is built into Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS). This switch uses GRS to look at various route patterns and preferences and decide which preference is most appropriate at the time. With GRS, voice and data calls can be sent along separate routes or be integrated on the same trunk group. If the type of transmission is incompatible with the endpoint (for example, a digital data call is sent to an analog phone), GRS provides a conversion resource such as a modem from a modem pool to complete the call.

Detailed description

GRS recognizes one or more Bearer Capability Class (BCC) for each trunk group preference in the route pattern. BCC defines the type of information being sent as voice or data. The switch checks the BCC for all trunk groups to see if the route selected and type of call are compatible. The BCC is assigned to the route preference on the Route Pattern screen.

GRS chooses a preference with BCC set to yes in this order: BCC 2, BCC 1, BCC 3, BCC 4.

When an exact match is not found in any route-pattern preference, calls with originating BCCs listed are treated as follows:

- BCC of 0 (such as voice or analog modem)

  GRS routes a BCC 0-originated call with no match. This allows voice transfer to data when making a data call.

  Since BCC 0 (voice) has no Information Transfer Capability (ITC), the switch selects an ITC from the route pattern when a BCC 0 call is routed as a data call. Table 61 shows how the ITC codepoint in the Bearer Capability IE is determined.

Table 61. Determination of ITC Codepoint

Originating Endpoint's ITC	Routing Preference's ITC				ITC codepoint in BC IE
	restricted	un-restricted	both endpoint	both un-restricted	
voice	x				restricted
voice		x			unrestricted
voice			x		unrestricted
voice				x	unrestricted

- BCC 2

  If there is no preference with BCC 2 yes, GRS chooses a preference with BCC 0 yes. If BCC 0 yes does not exist, the call is blocked.

- BCC 1, 3, or 4

  BCC 4 (DCP/DMI Mode 0), BCC 1 (Mode 1), and BCC 3 (Mode 3) calls requires an exact match in order for the call to complete. ITCs must also match.

Example

Assume a route pattern is set up with BCC 0 and BCC 2 set to yes in preference 1, and BCC 1, BCC 3, and BCC 4 set to yes in preference 2.

A voice or Mode 2 data call accessing this pattern uses preference 1. A Mode 1, Mode 3, or Mode 0 data call uses preference 2, regardless of what trunks are available in the first preference.

Data modules and GRS

For all endpoints, the switch automatically determines its current operating mode when a data module begins operations. The default is Mode 2.

Because call origination from a data module determines the mode used on the call, you should press the Originate/Disconnect button if you change data options. This way, the right mode is assigned to the next call.

Table 62 lists the BCC for different types of information and endpoints.

Table 62. BCC Assignment

Endpoint	Voice/Data Mode	BCC
Phone	Voice	0
Data Line Circuit Pack	2	2
Voice Data Set	2	2
Modular Processor Data Module	0,1,2	1,2,4
Modular Processor Data Module-M1 (For ACCUNET Switched 56 kbps Service)	1	1
Modular Trunk /Data Module	2	2
Digital Terminal/ Data Module	2	2
510D Personal Terminal	2	2
Digital Communications Protocol Interface	0,2,3	2,3,4
7400A Data Module	2	2
3270T Data Module	3	3
3270C Data Module	3	3
3270A Data Module	2,3	2,3

Legend

BCC	Type	DCP/DMI Mode
0	Voice-Grade Data and Voice	None
1	56 kbps Data (Mode 1)	1
2	64 kbps Data (Mode 2)	2
3	64 kbps Data (Mode 3)	3
4	64 kbps Data (Mode 0)	0

Related topics

Refer to ''Route Pattern'' on page 1039 for information on how to set up route patterns.

Group paging

Group paging allows users to make an announcement over a group of digital speakerphones.

- You can create up to 32 paging groups on one Avaya DEFINITY Server.
- Each group can have up to 32 extensions in it.
- It's OK to assign the same extension to different groups.

Brief description

You, the switch administrator, create paging groups and assign extensions as members to the appropriate groups. Each group is assigned its own identifying extension, and users page the group by dialing this extension. When a user dials the group's extension, the switch activates the speakers on all the phones in the group. Speakerphone paging is one-way communication: group members hear the person placing the page but cannot respond directly.

Restrictions

Pages aren't always heard on every phone in a group. An extension does not transmit a group page if it has an active or ringing call or if it is off-hook. Listeners may drop a page by disconnecting. Refer to ''Interactions'' on page 1571 for features that block group pages.

Controlling access to paging groups

Each paging group is assigned a class of restriction, so you can provide or deny access to different classes of users by setting calling permissions appropriately. Note that you can administer classes of restriction so remote callers can make speakerphone pages. If you don’t want to allow remote users to page, you may want to set calling permissions (on the Class of Restriction screen) for VDNs and trunk groups so that neither can initiate pages.

Interactions

- Attendant Intrusion

  Attendants cannot intrude on group pages. If the attendant tries to intrude on the paging originator, the intrusion attempt succeeds. However, all group page members are able to hear both the paging originator and the attendant.

- Auto Exclusion and Manual Exclusion

  Bridged appearances are not allowed on the page. Therefore, the Auto Exclusion and Manual Exclusion features are disabled. Auto Exclusion is not activated because there are no bridged appearances to alert when the page terminates.

- Auto Hold

  Auto Hold does not put a group page on hold.The page is dropped and the incoming call is answered.

- Automatic Callback

  Automatic Callback is disabled when calling an active page group.

- Bridging

  Bridging is disabled on this feature. A bridged appearance of a group member does not receive any indication of a call when the page arrives. The bridged appearance cannot bridge onto the page.

- Call Coverage

  Pages do not follow group members’ coverage paths. A page group cannot be a coverage point.

- Call Park

  Group members who receive a page cannot park the call.

- Call Pickup/Direct Call Pickup

  Other extensions cannot pick up a group page.

- Call Forwarding

  Group pages cannot be forwarded.

- Conference

  Neither group members receiving a page nor the originator of the page can conference the page to other extensions.

- Distributed Communications System (DCS)

  Page groups cannot be administered across DCS switches. DCS is not supported.

- Do Not Disturb

  If a member of a page group activates Do Not Disturb, that member does not receive pages.

- Go to Cover

  The Go to Cover feature is ignored because group pages do not follow coverage.

- Hold

  The originator of a group page can put the page on hold, but group members cannot.

- Leave Word Calling

  Leave Word Calling (LWC) is disabled. A page group cannot receive messages.

- Manual Signaling

  The Manual Signaling feature cannot be assigned to a page group.

- Send All Calls

  If a member of a page group activates Send All Calls (SAC), that member does not receive pages.

- Service Observing

  Group page members and page originators cannot be observed while active on a page.

- Transfer

  Group members cannot transfer a page.

- Trunks

  Trunks cannot be added to a page group.

- Vectoring

  Paging groups cannot be explicitly added to a vector path.

  NOTE:
  If a vector has a collect digits step and a route-to digits step, a person who uses the vector can enter a page group extension. Ensure that the COR of the vector restricts the vector from calling the page group if this action is not desired.

Related topics

Refer to ''Paging over speakerphones'' on page 469 to administer group paging.

Hospitality features

This section describes the following Avaya MultiVantage features that are tailored to hospitality applications:

- ''Attendant Room Status'' on page 1573
- ''Automatic Selection of DID Numbers to Guest Rooms'' on page 1574
- ''Automatic Wakeup'' on page 1575
- **''Custom Selection of VIP DID Numbers''**
- ''Do Not Disturb'' on page 1581
- ''Names Registration'' on page 1584
- ''Property Management System Interface'' on page 1587
- ''Suite Check-in'' on page 1594

Each feature indicates how to administer the Attendant Console screen and Hospitality screen to enable the hospitality features.

Attendant Room Status

Attendant Room Status allows the attendant to see whether a room is occupied and each room's housekeeping status.

NOTE:
This feature is available only if you have Enhanced Hospitality enabled on the System-Parameters Customer-Options screen and you have the DXS lamp field on the console. (Contact your Avaya account representative for information.)

Check In/Check Out Status

You can allow the attendant to review the check-in/check-out status by assigning an occ-rooms (occupied rooms) button on the Attendant Console screen.

When the attendant activates check-in/check-out mode, the DXS lamps light for every occupied room.

Maid Status

You can allow the attendant to review the maid status by assigning a maid-stat button on the Attendant Console screen.

When the attendant activates the maid status mode, the system prompts the attendant to enter the room status number (1 to 6) that they want to review. You can define these six room states on the Hospitality screen. Once they enter a room state, the display shows the definition of the room state and lights the DXS lamps for every room in that state.

While the console is in maid status mode, the attendant can review another room state by entering the room status number.

NOTE:
The attendant cannot make outgoing calls via the keypad while the console is in maid status mode; they must return to normal mode.

Automatic Selection of DID Numbers to Guest Rooms

Automatic Selection of Direct Inward Dialing (DID) Numbers for Guest Rooms allows you to give guests, upon check-in, phone numbers that provide direct dial access to their room. The switch automatically chooses a number from a rotating list of available DID numbers to be assigned to a guest's room. This provides a measure of privacy to your guests because providing the phone number does not give away the room number.

Callers would use a 7- to 10-digit number from outside of the hotel. For calls from inside the hotel, callers would use either the room/extension number or the 2- to 5-digit DID number.

For example, when a check-in is done from the switch (via the CHECK-IN button on the console) or remotely via a Property Management System (PMS) system, the switch assigns a DID number to the checked-in room from a list that is assigned at the switch. All calls made to the DID number are directed to the room as if the room was called directly.

NOTE:
The following process presumes you have established a dial plan and administered all DID numbers to their extensions (on the Station screen) as XDID station types.

Interactions

- Coverage

  XDID ports perform hunt-to before coverage. After hunting, coverage criteria for these calls is based upon the Direct Inward Dialing (DID), but the coverage points are based upon the hunted-to phone (room).

- Coverage

  Do not assign a COS with Client Room enabled for the XDID station types.

Automatic Wakeup

Automatic Wakeup allows attendants, front desk users, and guests to request an automatic wakeup call at a later time.

If the Dual Wakeup field on the Hospitality screen is **y**, each extension is allowed two wakeup call requests within one 24-hour time period. If the Room Activated Wakeup with Tones field is **y**, wakeup calls can be activated via tones that prompt users for the time they want to be called.

Detailed description

Wakeup requests may be placed from 5 minutes to 23 hours and 55 minutes in advance of a wakeup call.

Depending on how automatic wakeup is administered, when a user answers a wakeup call, the system can provide:

- a recorded announcement
- a speech-synthesis announcement
- music
- silence

All wakeup times entered into the system round to the nearest five minutes. For example, a requested time of 6:58 am stores in the system as 7:00 am. The switch bases time-validity checks on the rounded figure.

Wakeup calls are placed within two and one-half minutes of the requested time, and never reroute, forward, or go to coverage. Before placing the wakeup call, the system overrides Do Not Disturb for the extension.

If a wakeup-call attempt is not answered or if the extension is busy, the system tries two more times at 5-minute intervals. If the call does not complete after 3 attempts, the switch leaves a LWC message for a designated extension (usually assigned to a button on the attendant console or backup phone). The system maintains a complete record of all wakeup-call activity for the past 24 hours.

Users with touch-tone dialing can enter a wakeup request (if they have a speech synthesizer circuit pack and no display set or if Wakeup Activation via Tone is enabled) or can have the front desk set a wakeup time. Users with rotary-dial phones call the front desk to request a wakeup call.

Activate Automatic Wakeup either by dialing the FAC or by pressing the automatic wakeup entry button. If the system has a speech synthesizer circuit pack, the system provides voice prompting. If the user has a display set, the system provides display prompting.

- Voice Prompting with Room Activated with Tones Off

  A guest enters his or her own wakeup-call request. The request is entered only for the extension where the call originates.

  After the user dials the Automatic Wakeup FAC, the system generates voice prompts (the system must have a voice synthesizer circuit pack). These prompts tell the user when to enter information and what information is needed. Use touch-tone buttons to enter the information. The system accepts 24-hour or standard time. The user dials the automatic wakeup FAC again to change or delete a wakeup request.

  If the user makes invalid entries, a standard message generates that notifies the user of the error. The system then repeats the original prompt for input. If invalid entries occur on the second try, the system informs the user to dial the attendant for assistance.

- Voice Prompting with Room Activated with Tones On

  A guest enters his or her own wakeup-call request. The request is entered only for the extension where the call originates.

  After the user dials the Automatic Wakeup FAC, the system generates recall dial tone (the system does not need a voice synthesizer). This dial tone prompts the user to enter the time in a 24-hour, 4-digit format. Confirmation tone means that the wakeup request is successful.

- Display Prompting with Dual Wakeup Off

  Display prompting is provided to attendants, front-desk users, and other users with display-equipped phones. Administer front-desk users (or any other phones you want to grant permissions to) with a console permission class of service (COS) to perform the same actions as the attendant. Other users can enter a wakeup request only for the extension where the call originates.

The attendant presses the automatic wakeup entry button to activate the feature. If the attendant is on an active call with a system user, the user's extension displays as the default extension after pressing the pound sign (**#**). If the displayed extension is not the extension of the user requesting the wakeup call, the attendant can change it. Display prompting continues until the attendant enters all necessary information and the request for the wakeup call is confirmed.

If a condition exists that does not allow the system to accept the wakeup request, the system displays the reason for denial. Wakeup requests are denied for one of the following reasons:

- — Too Soon — Indicates that the requested wakeup time is within the current five-minute wakeup interval
- — System Full — Indicates that the maximum number of wakeup calls is reached
- — Interval Full — Indicates that the maximum number of wakeup calls in any 15-minute interval is reached

The attendant can change or cancel a wakeup call request at any time.

- Display Prompting with Dual Wakeup On

  Display prompting with Dual Wakeup works the same as Display Prompting with Dual Wakeup off (described in the previous text), except that after the first wakeup request is entered, the user is prompted for the second wakeup request.

When the system places a wakeup call, one of the following occurs:

- — Extension Is Busy — The wakeup call is placed again later.
- — No Answer — The extension rings for 30 seconds. If the call is not answered, the system tries again later.
- — Ringing Blockage — If four or more ports on the same analog-circuit pack are already ringing, the system waits 16 seconds and tries again. If the second attempt is blocked, the call has failed and the system waits 5 minutes before trying again.
- — Call Is Answered — The guest answers the wakeup call and hears either music, a recorded announcement, the speech-synthesizer announcement, or silence.
- — System Reset — indicates that a system reset level 1 or system reset level 2 occurred while the system attempted to place the wakeup call. Calls affected by these conditions are treated as other wakeup attempts.

If a wakeup call is incomplete because of a busy, no answer, ringing blockage, or system reset, the system attempts to place the call 2 more times at 5-minute intervals. If the call is not completed after 3 attempts, the system leaves an LWC message to account for the failed attempt.

A special extension, called the Extension to Receive Failed Wakeup LWC Messages, is administered exclusively for receiving failed wakeup-call LWC messages. When a failed message is retrieved, the display shows the date, time, and extension for the failed wakeup-call attempt.

Assign an automatic-message waiting (AMW) button and associated lamp to attendant consoles or front-desk terminals. The number associated with the button can be the wakeup-messages extension. The AMW lamp lights when a failed wakeup message is waiting. The user retrieves the message by invoking coverage-message retrieval on the wakeup-message extension. The user presses the AMW button to place the console or phone in coverage-retrieval mode. The user then retrieves the failed wakeup-call attempt messages. Only attendants and specified phone users can retrieve and delete failed wakeup messages.

The system maintains an audit-trail record of wakeup-call activity for the past 24 hours. The wakeup-call buffer can only hold a number of records equal to the maximum number of stations administrable on the switch. For example, if a maximum of 200 stations is administrable, only 200 automatic-wakeup records are stored.

You can display wakeup events at the management terminal, or print to a designated printer. If the system has a journal printer, wakeup events print as they occur.

The audit trail record contains the following information:

- Type of event:
  - Request — A new wakeup-call request is made.
  - Change — The time is changed on an existing wakeup-call request.
  - Cancel — A wakeup request is canceled.
  - Move To — The wakeup request for this room moves to another room.
  - Move From — The wakeup request for another room moves from the old room to the new room.
  - Move-Cancel — A wakeup request from another room replaces the request for this room.

  - Swap — A room swap occurs and at least one of the rooms has a wakeup request. Wakeup calls swap when a room swap is performed. A journal entry is made for each room. If the room receives a wakeup call as the result of the swap, the time of the call is provided in the entry. If the room loses a wakeup call as the result of the swap (and has not received another), the time is not present in the entry.
  - Completed — The wakeup call completes successfully.
  - Not Completed — The wakeup call failed.
  - Skip — The wakeup call is skipped. This event occurs if the system time advances past the requested time of a wakeup call.
- Time of the event
- Extension number receiving the call
- Time of the wakeup request
- Extension (or 0 for the attendant) where the event took place
- Number of call attempts that were placed
- An indication of why a wakeup-call attempt failed

In addition, all wakeup-time changes are recorded. This record shows the original time requested and the changed time. The audit-trail record is not backed up and all wakeup data is lost if a system failure occurs.

Schedule the following reports for printing on a daily basis:

- Wakeup Activity report – summarizes wakeup activity for each extension that had any wakeup activity over the past 24 hours.
- Wakeup Summary report — gives an hour-by-hour summary of the number of scheduled wakeup calls, the number of wakeup calls completed, and a list of extensions. The report covers all automatic-wakeup events for each hour over a 24-hour period.

With vector directory numbers (VDNs) and multiple announcements, you can choose as the announcement extension a VDN that reaches one announcement if the system clock is less than 12:00 and another if the system clock is greater than 12:00. The hotel guest hears “good morning” before noon and “good evening” after noon. Or, a business customer can choose as the announcement extension a VDN that points to an extension assigned to a quorum bridge, with the wakeup time as a scheduled teleconference time. When the wakeup call is completed, the customer automatically connects to the teleconference bridge.

You can administer a multiple announcement to repeat. To enable repeating announcements, enter announcement type **integ-rep** command on the Recorded Announcement screen. With repeating integrated-message functionality, the announcement keeps repeating from when the first guest (of a group of guests receiving the same wakeup announcement at the same time) goes off-hook until the last guest goes on-hook.

If the announcement type is either an externally-recorded announcement or is integrated-repeating, you can administer the wakeup-call queue for barge-in. Barge-in means that the guest receiving the wakeup call hears the announcement as soon as he or she is off-hook, even if the announcement is not at the beginning. This provides the capability of many users being bridged onto the same announcement port, eliminating the need for a separate port for each wakeup call. Refer to ''Recording announcements'' on page 434 for additional information.

Considerations

- Up to 10 attendant consoles and/or front desk terminals may be in the wakeup display mode at any one time.
- Wakeup call attempts are not rerouted, forwarded, or sent to coverage.

Interactions

- Attendant or Phone Display

  If the console or phone is in automatic-wakeup mode and the user presses another display-mode button, wakeup mode aborts and the wakeup request is not entered, changed, or deleted.
- Do Not Disturb

  If Do Not Disturb is active at a phone, Automatic Wakeup deactivates Do Not Disturb for that terminal, and the system places the wakeup call.
- PMS Interface

  A Check-Out request cancels an active-wakeup call request for the guest room. Room Change/Room Swap requests through PMS cause a wakeup request to change or swap.
- Speech Synthesizer Circuit Pack

  Auto Wakeup competes with the following features for use of the speech-synthesizer circuit pack.

  — Do Not Disturb

  — Leave Word Calling Message Retrieval

  — Visually Impaired Attendant Service

  If the Wakeup Activation via Tone is enabled, the auto wakeup interface from the Speech Synthesizer circuit pack is disabled. The Do Not Disturb interface still operates.

Custom Selection of VIP DID Numbers

Custom Selection of VIP DID numbers allows you to select the DID number assigned to a room when a guest checks in. It also provides buttons on display sets that allow you to check-in a VIP (the vip-chkin button), to view and change XDID and XDIDVIP numbers (the did-view button), and to disassociate XDID and XDIDVIP numbers outside of the normal guest check-out procedure (the did-remove button).

Make sure the following fields are administered in order to use Custom Selection of VIP DID Numbers:

- the Basic Hospitality field on the Feature Related System Parameters Customer Options screen is y
- the Custom Selection of VIP DID Numbers field on the Feature Related System Parameters Hospitality screen is y
- the Automatic Selection of DID Numbers field on the Feature Related System Parameters Hospitality screen is y

You also need to set up a number of stations as xdidvip (enter **xdidvip** in the Type field on the Station screen).

When you use the vip-chkin button on a display phone to check in a guest, you receive prompts to enter the room extension number and the VIP DID number. Use the did-view button to change a DID number that is automatically assigned by the switch (XDID), or one you select yourself (XDIDVIP).

Use **list station** to see which VIP DID numbers are administered. Check the hunt-to station field to see if an XDIDVIP number is available or is assigned to a guest room.

Do Not Disturb

Do Not Disturb allows guests, attendants, and authorized front-desk phone users (those with console permission) to request that no calls, other than priority calls, terminate at a particular extension until a specified time. At the specified time, the system automatically deactivates the feature and allows calls to terminate normally at the extension.

Do Not Disturb is a form of termination restriction associated with an automatic deactivate time. When Do Not Disturb is active, the user receives only those calls associated with Automatic Callback, Automatic Wakeup, and Priority Calling, and those calls that are redirected to that extension via the Call Coverage and Call Forwarding All Calls. All other calls redirect to a recorded announcement, an attendant, or intercept tone. You can administer the switch to provide a special dial tone whenever an analog set goes off-hook when Do Not Disturb is active.

Phone users with touch-tone dialing can activate this feature themselves or ask the front desk to do it for them. Users with rotary-dial phones must call the attendant or front-desk user to request Do Not Disturb.

Activation by phone users

Phone users can activate Do Not Disturb by dial access or by button access. If users have a speech-synthesizer circuit pack, they can activate Do Not Disturb themselves, without attendant assistance.

- Dial Access

  When a user dials a Do Not Disturb FAC, the system prompts the user to enter a deactivate time. The user may later change or delete the request by dialing the Do Not Disturb FAC again and entering the required information.

  If the user makes invalid entries or if system conditions prevent entry of the request, the system informs the user to dial the attendant or front desk for assistance, if the user has a speech-synthesizer circuit pack.

- Button Access

  If a phone has a Do Not Disturb button, the user can press the button to activate the feature. The handset may be on-hook or off-hook. The user presses the button a second time to deactivate the feature.

  The lamp associated with the Do Not Disturb button lights until the feature is deactivated with the button. An automatic-deactivate time is not provided.

Activation by Attendant

The attendant can activate the feature for a user or a group of users. (The assigned class of restriction (COR) determines which users are in the group.) The attendant presses the Do Not Disturb — Extension button followed by the extension, or the Do Not Disturb — Group button. The extension followed by the appropriate COR number.

The attendant can cancel a Do Not Disturb request by activating the feature, entering the desired extension or group COR number, and pressing the delete button.

Activation via a PMS

The system provides an interface to a Property Management System (PMS). This interface allows activation and deactivation of controlled restrictions. Activation of Do Not Disturb through a PMS is similar to activation of termination restriction. A scheduled deactivate time cannot be specified.

Audit Trail Reports

The system keeps a record of all phones that are in Do Not Disturb mode. You can display or print this information.

Administer the following reports for printing on a daily basis:

- Do Not Disturb Status Report — This report lists all extensions with Do Not Disturb active and the specified deactivate time for each.
- Do Not Disturb Plus COR Status Report — This report lists all extensions, plus those whose controlled-restriction level is termination restriction. (The attendant activates termination restriction for a specific extension or COR. A deactivate time is not associated with termination restriction.)

Records do not include Do Not Disturb information for extensions that are both termination and outward restricted.

Considerations

- Do Not Disturb lessens the attendant's workload when phone users with speech-synthesizer circuit packs activate the feature themselves.
- A front-desk user must have a console-permission class of service (COS) to activate this feature.
- The number of available speech-synthesis ports is the only limit on the number of users receiving voice prompting.

Interactions

- Automatic Callback

  Do Not Disturb does not block an Automatic Callback call. Return calls terminate at a phone in the normal way.

- Automatic Wakeup

  An Automatic Wakeup call deactivates Do Not Disturb and alerts the guest at the specified time.

  If the Wakeup Activation via Tone is enabled, the auto wakeup interface from the Speech Synthesizer circuit pack is disabled. The Do Not Disturb interface still operates.

- Call Coverage

  If a point in a coverage path has Do Not Disturb active, calls covering to that extension alert the extension unless the extension has controlled-restriction termination active. When Do Not Disturb is active and a phone does not have a coverage path, calls are routed to the attendant.

- Call Forwarding All Calls

  If Do Not Disturb is active at the forwarding extension, the caller receives intercept treatment. If Do Not Disturb is active at the forwarded-to extension, the call alerts the forwarded-to extension.

- Controlled Restriction

  When a phone has total-controlled restriction, it cannot receive or place any calls. However, it can receive a call if another station has an auto-icom button pointing to the controlled-restriction station.

- Internal Automatic Answer (IAA)

  Activation of Do Not Disturb at the called phone preempts IAA.

- PC Console

  You cannot implement Do Not Disturb at a PC Console.

- PMS Interface

Checkout from either a PMS or the switch automatically deactivates Do Not Disturb for the specified extension.

Names Registration

Names Registration automatically sends a guest’s name and room extension from the PMS to the switch at check-in, and automatically removes this information at checkout.

Detailed description

The information provided by Names Registration displays on any attendant console or display-equipped phone (as might be used for example, by Room Service, Security, and others). The information allows hotel personnel to provide personalized greetings to calling guests. For example, if John Smith calls room service, personnel with a display-equipped phone, see John’s name and room extension and can answer with a personalized greeting.

The name of the calling or called party can display on display-equipped phones. To maintain necessary guest security, hotels do not divulge guests’ room numbers to other guests or callers. For this reason, do not assign display-equipped phones to guest rooms.

Check In

The switch performs the following procedures at check-in:

1. Information about the guest is obtained and stored in the hotel’s PMS.
2. The PMS sends a check-in message to the switch.

3. The switch stores the guest's name and coverage path.
4. The switch removes the outward restriction on the telephone in the guest room. The switch removes all LWC messages.
5. The switch changes the status of the room from unoccupied to occupied.

   At check-in, update the PBX names internal table and the call-coverage path for the guest phone. Name Registration automatically sends a guest's name, extension (room), and preferred call-coverage path to the switch.

Check Out

1. The switch clears any previous wakeup calls.
2. The switch clears message-waiting lamp indications.
3. The switch activates controlled outward restriction, removes the guest's name, and identifies any unopened messages.

At checkout, Name Registration automatically changes the call-coverage path to the administered Default Coverage Path for Client Rooms.

Guest Information Input/Change

Use Guest Information Input/Change to change the guest name associated with an extension, input a guest name after check-in, or change a call-coverage path. For example, hotel may check in airline personnel before their arrival to guarantee their reservation. However, hotel personnel may be unaware of the guests' names and so wait until their arrival to update the names.

Name Registration Information Format

For both Name Registration and Guest Information Input/Change, a guest name may consist of as many as 15 characters, including spaces and commas. Do not use periods.

The name may be in all upper case letters, all lower case letters, or a mixture of upper case and lower case letters. To use Integrated Directory, enter the name using one of the following methods.

- Last name, comma, first name (for example, Jones, Fred)
- Last name, comma, first name, space, title/middle initial/name (for example, Jones, Fred Mr)
- Last name only (for example, Jones)

Call Coverage

Both Names Registration and Guest Information Input/Change messages contain call-coverage path numbers. These numbers do not display but are used to configure the appropriate call-coverage arrangements for guest extensions. Arrangements can be for voice mail, text messages, any available coverage point, or no coverage at all.

Administer call-coverage paths on the switch, and use the associated path numbers to establish coverage arrangements at check-in. For suites, administer paths to allow one room in the suite to be the coverage point for the other. To make customized arrangements at time of check-in (such as coverage from one guest room to another), manually administer the path attributes at the switch.

Considerations

- Call-coverage path numbers sent by PMS to the switch for automatic reconfiguration are limited to those administered in the switch and stored in PMS.
- A guest room extension can have a maximum of 5 digits.
- An input in PMS of the name displayed on display-equipped phones updates the switch.

Interactions

- Call Coverage

  Call-coverage arrangements are not limited to automatic update during check-in messages sent from PMS. Hotel personnel require coverage points other than those designated for guests. Call-coverage paths can be manually administered at the switch via the management terminal.
- COS

  If an extension has a client room COS, the save translation operation clears the station name and sets the coverage path to the default coverage path for client room when stored on tape. This does not affect the existing information in memory. However, if the translations are read in, it affects existing extensions until a database swap synchronizes the switch and PMS.
- Name Character Length

  The switch supports 27-character names, but the PMS interface supports only 15-character names.
- PMS Interface

  During a Room Change/Room Swap, the name originally associated with the first terminal is changed or swapped to the second terminal along with call-coverage path, automatic wake-up entries, message-waiting status, and controlled restrictions.

Property Management System Interface

Property Management System (PMS) Interface provides a communications link between the switch and a customer-owned PMS. The PMS allows a customer to control certain features in a hospital and hotel/motel environments. Refer to *Avaya MultiVantage running on a DEFINITY GuestWorks Server Property Management Interface Specifications*.

Detailed description

Table 63 summarizes how the hospitality features are activated when you use only the switch and when you use the PMS.

Table 63. PMS/Switch links

Feature	Switch Only	With PMS
Automatic Wakeup	Activated via console button	N/A
Call Coverage	Activated via administration	Activated via PMS terminal — Transparent or ASCII mode
Check-In/Check-out	Activated via console button	Activated via PMS terminal — Normal, Transparent, or ASCII mode
Controlled Restriction	Activated via console button	Activated via PMS terminal — Normal, Transparent, or ASCII mode
Do Not Disturb	Activated via console button	Activated via PMS terminal — Normal, Transparent, or ASCII mode
Emergency Access to Attendant	Activated by guest action	N/A
Housekeeping Status	Activated via console button	Activated via PMS terminal — Normal, Transparent, or ASCII mode

Continued on next page

Table 63. PMS/Switch links (Continued)

Feature	Switch Only	With PMS
Message Waiting Notification	Activated via console button	Activated via PMS terminal — Normal, Transparent, or ASCII mode
Names Registration	Activated via administration	Activated via PMS terminal — Transparent or ASCII mode
Room Change/Swap and Guest Information Input/Change	Activated via administration	Activated via PMS terminal — Normal, Transparent, or ASCII mode
Room Occupancy	Activated via console button	Activated via PMS terminal — Normal, Transparent, or ASCII mode

The PMS Interface provides the following:

- A communications protocol for controlling message exchange between the switch and a PMS
- An application module for controlling the operation of PMS features
- Status data on all guest/patient rooms for selected features

The protocol is full-duplex asynchronous and provides the mechanisms for setting up a data session with PMS, message-exchange control, error identification, and recovery. The interface supports standard data rates.

Two protocol modes are provided: the normal-protocol mode as described above, and transparent-protocol mode. Normal-protocol mode supports a character set that has a combination of Binary Coded Decimal (BCD) characters and ASCII characters. Transparent-protocol mode supports a complete ASCII-character set.

The application module of the PMS Interface implements requested features and provides backup if the PMS link is down. Whether or not the link is down, the switch always maintains the following data for each room:

- Whether the room is vacant or occupied
- Whether the phone's message lamp is on or off
- Whether a controlled restriction is active at the phone and, if so, which one
- The guest's name and coverage path

When the PMS link is down, the switch automatically activates Check-In/Check-Out for the attendant console and front-desk terminal with display capability, and continues to support PMS features activated from guest/patient-room phones.

When the PMS link is up again, the switch sends one of the following messages to PMS:

- No room-status changes occurred during loss of communications.
- Room-status changes did occur during loss of communications; therefore, a data exchange is needed to synchronize the switch and the PMS databases.
- The system failed momentarily, destroying its record of room status; therefore, a data exchange is needed to synchronize the switch and the PMS databases.

When the PMS link is down or not used, the switch maintains an audit-trail report of all events that are normally sent to the PMS. The audit-trail report (accessed via the management terminal) is a sequential listing of all PMS transactions executed by the switch when the PMS link is down. Included are error events that occur when the link is up or down.

If you have a PMS printer and the PMS link is down, the following status changes print as changes occur:

- Room number
- FAC dialed
- Any additional information digits that were dialed
- Reason for the entry (error message)
- Time that the error occurred

Additional reports print to the PMS Journal/Schedule printer. These include Automatic Wakeup activity, Emergency Access to the Attendant activity, and scheduled reports.

A supporting function called Room Data Image synchronizes the switch and PMS databases after a PMS link goes down and comes back up. Information exchanged includes:

- Room extension
- Whether the room is occupied or vacant
- Message Waiting lamp status
- Controlled Restriction status
- Guest’s name
- Call Coverage path

Message Waiting Notification

Message Waiting Notification requests originate from attendant consoles, front-desk terminals, or PMS terminals. When a request is entered, PMS sends a message to the switch to change the state of the Message Waiting lamp. If the lamp is activated by AUDIX, INTUITY Lodging, or Leave Word Calling (LWC), the PMS cannot deactivate the lamp. PMS cannot turn LWC or AUDIC messages on or off; these are controlled by the switch.

Assign a console permissions COS to any console or terminal as part of the “System Wide Retrieval Stations” to retrieve requests for another station. Assign a client room COS to the extensions for which Message Notification is to be made.

Controlled Restriction

When Controlled Restriction is activated through the PMS, the PMS sends a message to the switch to assign one of the following restrictions to the phone in a guest/patient room:

- No restriction
- Outward restriction
- Total restriction
- Station-to-station restriction
- Termination restriction
- Combined outward and termination restriction
- Combined outward and station-to-station restriction
- Combined termination and station-to-station restriction

The attendant can still set Controlled Restriction for a phone whether the PMS link is up or down.

PMS-Down Log

The pms-down log records only those User Controller Restriction events that are for stations having a Class of Service (COS) where:

- the Client Room field is **y**
- the Controlled Restriction Configuration field is **act-pms**
- the pms link is not up
- the pms log extension is valid

Housekeeping Status

Your housekeeping staff enters status information from phones in guest/patient rooms or from designated terminals. You can assign up to 10 Housekeeping Status access codes within two different types:

- Room phone access code type

  Staff members dial up to six access codes that represent room status plus up to six additional digits for items such as maid identification.

- Designated phone access code type

  Staff members dial up to four access codes that represent room status plus the room extension and then up to six additional digits for items such as maid identification.

The switch notifies PMS when Housekeeping Status information is entered. If the PMS is unavailable, the switch writes this information to a log. The log is accessible at the switch management terminal, and is sent to the log printer, if administered.

Check In/Check Out

A Check-In request deactivates the outward-controlled restriction on the phone in a guest/patient room. A Check-Out request deactivates any controlled restrictions and changes the controlled-restriction level to outward restriction, checks for any messages, clears the wakeup request, and deactivates Do Not Disturb.

If you do not use PMS or if the PMS link is down, the attendant can activate Check-In and Check-Out from an attendant console or a front-desk phone with display capability and console permission. This requires two buttons, Check-In and Check-Out. Pressing either button places the display in the respective mode and allows use of the touch-tone or DTMF buttons for entering data (rather than for placing calls).

The attendant exits Check-In or Check-Out mode by pressing any other button associated with the display (for example, the Normal Mode button). This restores the display and the touch-tone or DTMF buttons to normal operation.

A Check-In/Check-Out request sends information for Names Registration to the switch. This information includes the guest's name, room extension, and call-coverage path. If the PMS link is down and check-in is done from an attendant console or display-equipped front-desk phone, the guest's name and coverage-path information is not automatically updated.

If a guest/patient room has both a voice and a data extension, the checkout request applies only to the voice extension.

Room Change/Room Swap

Room Change/Room Swap is provided only through PMS and activated from a PMS terminal. With Room Change, data pertaining to the old room — including a pending wakeup request, the guest's name (transparent/ASCII mode), and the guest's call-coverage path (transparent/ASCII mode) — moves to the new room. With Room Swap, data pertaining to the two rooms swap. With either feature, if the occupancy status is inconsistent, the system sends an error message to PMS.

Names Registration

Names Registration automatically sends a guest's name and room extension from PMS to the switch at check-in, and removes this information at checkout. The guest's call-coverage path is sent to the switch during check-in and set to the administered Default Call Coverage Path for Client Rooms at checkout.

Guest Information Input/Change

Guest Information Input/Change allows the attendant to enter or alter guest information (name or coverage path). Information changed at the PMS is sent automatically to the switch.

PMS/INTUITY Link Integration

PMS/INTUITY Link Integration allows the following PMS administrative messages to tandem through the switch to the INTUITY Lodging adjunct. This eliminates the need for the INTUITY-to-PMS voice messaging link. This does not remove the need for the INTUITY-to-PMS call accounting link.

- Check-in
- Check-out
- Room-data-image (database synchronization)
- Modify (guest-information)
- Add/Remote Text/Fax Notification Message (message-waiting status)
- Transfer/Merge Mailbox (room change/swap)

When the messaging link is down and the PMS/MultiVantage link is up, the switch buffer holds up to 100 PMS messages. The switch updates the Intuity Lodging adjunct once the link is up. If the buffer overflows before the link is up, the database resync among PMS/MultiVantage/Intuity initiates by demand or by a routine database update from PMS.

Considerations

- You can use Leave Word Calling (LWC) or Integrated Message Center Service for the hospital or hotel/motel staff and Message Waiting Notification for guests/patients. However, if you do not use Message Waiting Notification, Integrated Message Center Service is used for both.
- Do not remove an extension while the PMS link is active.
- Normal-protocol mode allows extensions of up to four digits. Transparent/ASCII-protocol mode allows extensions of up to five digits.
- When save translations is done when transparent/ASCII-protocol mode is active, station names with client-room COS save as blank and coverage paths save as the default coverage path for client rooms.
- The PMS link may not work correctly when multiple p-extensions have the same leading digit and adjacent lengths. For example, 3 and 4 p-extensions with the same leading digit may cause problems. The same applies to 4 and 5, and 5 and 6.
- A room extension may begin with 0 only if the PMS sends a prefix digit or a fixed number of digits.

Interactions

- Attendant Console or Front Desk Terminal

  Activate Controlled Restriction, Check-In/Check-Out, and Message Waiting Notification at an attendant console or a front-desk phone with console permission. The attendant console receives visual notification of the status of the PMS link between the system and the PMS.
- AUDIX Interface

  Message lamps activated by this feature cannot deactivate with feature buttons or with feature messages from the PMS.
- Automatic Wakeup

  Set or cancel an Automatic Wakeup request for a guest room as a result of Room Change/Room Swap or Check-Out.

- Do Not Disturb

  Set or cancel a Do Not Disturb request for a guest room as a result of a different Controlled Restriction, Room Change/Room Swap, or Check-Out.

- Leave Word Calling (LWC)

  Message lamps activated by this feature cannot deactivate with Manual Message Waiting feature buttons.

  If Room Change is active, LWC messages for the old room do not move to the new room. If Room Swap is active, LWC messages for the two rooms do not swap. Therefore, do not encourage use of LWC in guest rooms.

- Restriction — Controlled

  Controlled Restriction for a group of user extensions, when activated from the switch, is not conveyed to the PMS. The PMS is not able to add or remove such restrictions by sending feature messages.

Suite Check-in

Suite Check-in allows you to have the switch automatically check in more than one phone with one check-in command (whether from your PMS or from the switch).

When a room phone is checked in, the switch looks for a hunt-to extension associated with that station. If it finds one, the switch also checks in the station found in the hunt-to field. The switch also:

- removes controlled outward restriction
- adds the guest’s name to the station record for that extension
- stores the call coverage path
- removes any Leave Word Calling (LWC) messages
- marks the room as “occupied”

If the hunt-to (second or subsequent) station has an extension in its hunt-to field, that station also is checked in. The switch continues checking in stations until it meets a station in the chain with a blank hunt-to field.

Interactions

- Automatic Selection of DID Numbers for Guest Rooms

  If Automatic Selection of DID numbers is active, then a DID number is assigned only to the initial extension (the one that appears in the check-in message), not to all of the hunt-to extensions.

- Dial By Name

  Since the secondary phones that are checked-in insert a "*" before the name, they do not appear when Dial By Name is used. However, the name (with the "*" in front of it) appears when the phone dials the attendant or another display set.

- Do Not Disturb

  When Do Not Disturb is activated for a phone, it is active for just that phone and not other phones in the hunt-to chain.

- Housekeeping Status Change

  When a room status feature access code is dialed, the room status is updated only for the extension from which the code was dialed (not the hunt-to phones as well). Housekeeping should be instructed to dial the room status changes from the primary phone.

Hunt Groups

A hunt group is a group of extensions that can handle multiple calls simultaneously to a single phone number. For each call to the phone number, the system hunts for an available extension in the group and connects the call to that extension.

A hunt group is especially useful when you expect a high number of calls to a particular phone number.   A hunt group might consist of people trained to handle calls on specific topics. For example, the group might be:

- A benefits department within your company
- A service department for products you sell
- A travel reservations service
- A pool of attendants

In addition, a hunt group might consist of a group of shared telecommunications facilities. For example, the group might be:

- A modem pool
- A group of data-line circuit ports
- A group of data modules

NOTE:
You may also assign Automatic Call Distribution (ACD) to a hunt group. In this case, the hunt group is known as an ACD split. Refer to *Avaya MultiVantage™ Call Center Software Release 11 Guide to ACD Call Centers* for more details about ACD splits.

Detailed description

The following sections describe how a hunt group works.

Hunting methods

The system uses one of two types of hunting method to distribute calls:

Direct department calling	The system hunts for an available extension in the hunt group, always starting with the first extension in the group. If the first extension is busy, the system checks the second extension. If the second extension is busy, the system checks the third, and so on. When an extension is available, the system rings that extension to connect the call.
Uniform call distribution	The system hunts for the extension that has been available for the longest time. The system then rings that extension to connect the call. This type of hunting provides the most equitable distribution of calls. *Also, this type of hunting is required for a modem pool, data-line circuit ports, and data modules*.
Circular	Enter **circ** when the call should be routed in a “round-robin” order. The order in which the participating extensions are administered is the order in which calls are directed. The switch keeps track of the last extension in the hunt group to which a call was connected. The next call terminating on the hunt group is offered to the next station in the circular list independent of how long that station has been idle. The switch does not start searching at the same place each time.

NOTE:
Expert Agent Selection uses uniform call distribution and expert agent distribution. Refer to *Avaya MultiVantage™ Call Center Software Release 11 Guide to ACD Call Centers*.

Hunt group queues

You can set up a queue for a hunt group. When all extensions in the group are busy, calls wait in queue for the next available extension. You determine how many calls can wait in queue by setting the queue length.

If all hunt-group members are unavailable or the queue is full, the system treats the call as follows:

- If the call is internal or is carried on a Direct Inward Dialing (DID), DS1, or tie trunk, the caller hears busy tone.
- If the call is on a central office trunk, the caller hears ringing, but gets no answer.
- If the hunt group has call coverage, the system sends the call to a coverage point.

Refer to ''How hunt group extensions become unavailable'' on page 1599.

Queue warning level

You can set up a queue warning level and an associated queue warning indicator lamp. When the queue reaches this level, the lamp lights and remains lit until the queue drops below this level. You can have one lamp for each hunt-group queue. Install the lamp so the members of the hunt group can see it.

Call coverage

You can set up call coverage for a hunt group. Then, if a hunt-group queue is full, the system sends new calls to the coverage point.

If a call goes into a hunt group queue, it stays in queue for the Coverage Don't Answer interval, then redirects to the coverage point. A call coverage point can be another hunt group.

Announcements

You can record and assign one delay announcement to each hunt-group queue. An announcement can be shared among hunt groups. Normally, the announcement should tell the caller to wait and say the call will be answered in the order in which it was received.

A call that connects to a delay announcement remains in queue while the announcement plays. If the call has not been answered by the time the announcement completes, the caller hears music (if provided) or silence. When the call begins ringing a member of the hunt group, the caller hears ringing.

Delay announcement interval

You also define for each hunt group a delay announcement interval. When a call enters the queue, the interval starts. This interval (0 to 99 seconds) indicates how long a call remains in queue before the call connects to a recorded announcement. If Call Coverage is provided, the Don't Answer interval (one to 99 ringing cycles) may also begin when the call enters the queue. After these intervals begin, one of the following occurs:

- If the Don't Answer interval expires before the delay announcement interval expires, the call redirects to coverage.
- If no coverage point is available to handle the call, the call remains in queue and may connect to the delay announcement.
- If the delay announcement interval expires before the Don't Answer interval, the call connects to a delay announcement. If the announcement is already in use, the delay announcement interval is reset.

This process continues until the call is answered, goes to coverage, connects to an announcement, or ends because the caller hangs up.

If you set the delay announcement interval to 0 seconds, a call automatically connects to the announcement. The result is a "forced first announcement." In this case, the call does not connect to a hunt-group member until after the announcement. The caller does not hear music.

If a call redirects to another hunt group via Call Coverage, the caller does not hear either hunt group's forced first announcement. However, the caller may hear the first or second announcement of the covering hunt group.

Analog, aux-trunk, or integrated announcements

Delay announcements may be analog, aux-trunk, or integrated (digital). For an analog or aux-trunk announcement, callers who enter the queue hear the associated announcement the next time the system plays it. Callers who enter the queue after the announcement begins do not hear it until it starts again. For an integrated announcement, multiple callers can be connected to the same announcement at different times, depending on the availability of ports. Refer to ''Recording announcements'' on page 434 for more information.

Example

Assume that a hunt group has the following parameters.

- Queue length is 10 calls.
- Queue warning level is 5 calls.
- Recorded announcement delay is 20 seconds.

All hunt-group members are busy. A call enters the queue as the fifth call, which causes the queue warning level lamp to light. Hunt-group members see the lamp and try to quickly complete their present calls. Meanwhile, the call waits in the queue for 20 seconds and hears the recorded announcement. When a hunt-group member becomes available, the first call in queue connects to that group member. The queue warning-level lamp turns off because the number of calls in queue fell to four.

How hunt group extensions become unavailable

An extension in a hunt group becomes unavailable to receive calls if the hunt group member is already handling a call. This rule is true even if the call is not a hunt-group call and even if the extension's phone is a multiappearance phone.

An extension also becomes unavailable if the member presses one of the following buttons:

- Hunt Group Busy
- Send All Calls
- Call Forwarding All Calls

  NOTE:
  If a member is also an ACD agent, pressing the AUX work button also makes the member unavailable. On the other hand, if an agent presses the ACW (after call work) button, the system considers the agent to be available and still will queue calls.

- Hunt Group member dials the Hunt Group Busy Activate feature access code

Hunt Group Busy

If you turn on the Hunt Group Busy option, a hunt group member can dial the Hunt Group Busy code followed by the hunt group number. The extension is unavailable for calls until the group member dials the Hunt Group Busy deactivation code or presses the button again.

If the last available member of a hunt group tries to activate the Hunt Group Busy option, the following occurs:

- New calls to the hunt group receive busy tone or go to coverage.
- Calls already in the queue continue to route to the last available extension.

- When the queue is empty, Hunt Group Busy activates. At the last available extension, the status lamp associated with the AUXILIARY WORK button, if provided, flashes until the queue is empty. When no more calls remain in the queue, Hunt Group Busy activates and the status lamp, if provided, lights steadily.

  NOTE:
  If an extension is an ACD split agent as well as a hunt-group member, the split agent normally has an AUX-WORK button that also activates/deactivates Hunt Group Busy. If an agent is the last available member and they push AUX-WORK, the button's light flashes until the queue is empty. This means the agent is still available. When the queue finally empties, the button lights steadily and Hunt Group Busy takes effect.

Send All Calls

If a station activates Send All Calls with the Send All Calls button, then calls to the hunt group extension go in queue if there is a queue, otherwise callers get a busy treatment if all the agents have their Send All Calls pushed.

NOTE:
If an extension is an ACD split agent as well as a hunt-group member, the split agent normally has an AUX-WORK button that also activates/deactivates Hunt Group Busy. If an agent presses the SEND ALL CALLS button, the agent becomes unavailable for hunt-group calls. The agent then can become available for calls again by pressing the SEND ALL CALLS button again.

Call Forwarding All Calls

With Call Forwarding All Calls active, an extension within a hunt group is unavailable for hunt-group calls. Callers hear the hunt group's forced first announcement, if administered, before the system forwards the call.

Considerations

- Members assigned to multiple hunt groups

  An extension can be a member of more than one hunt-group. However, a phone, even a multiappearance phone, can receive only one hunt group call at a time. On a multiappearance phone, all appearances must be idle to receive a hunt-group call.

  You can assign a COVERAGE INCOMING CALL INDICATOR (ICI) button to a multiappearance phone or attendant console. When a member receives a call for the hunt group associated with the ICI button, the button's status lamp lights.

- ACD agents as hunt group members

  Do not include ACD split agents in non-ACD hunt groups if they also receive ACD split calls. The system distributes all ACD calls to split agents before it distributes hunt-group calls.

  When you change an ACD split to a non-ACD hunt group, each split agent must enter the Hunt Group Busy deactivation code in order to receive calls in that hunt group. If the agent has an AUX-WORK button, the button lamp lights when you make the change. The agent can then press the button to become available for hunt-group calls.

- Hunt group for communications devices

  Members of a hunt group used for shared data communications must be of the same type. Thus, you can put data modules or analog modems in a hunt group, but not both. Option settings must be the same for all group members.

  A caller can still use the DATA EXTENSION button to access the associated data module, even if the module is in a hunt group. Individual data modules or modems can originate and receive calls.

- Access restrictions

  You can restrict, via the Class of Restrictions, any extension in a hunt group from receiving calls other than those to its assigned hunt group. You can also restrict extensions on your switch from calling the hunt group's extension.

- System limits

  The size of your system determines how many hunt groups you can set up and how many extensions you can assign to each group.

- Trunk signaling

  A hunt group always has its own extension. Therefore, a caller with a phone on the switch can call the hunt group by dialing only that extension. If a trunk group has the ability to pass digits from the CO to the switch (for example, a DS1 trunk group), a caller can also call the hunt group by dialing a 7-digit phone number that consists of a specified prefix and the hunt group's extension.

  If a trunk group cannot pass digits from the CO to the switch, incoming calls on that trunk group can connect to a hunt group only if the trunk group has the hunt-group extension as its primary destination. This includes trunk groups for incoming LDN calls, international exchange calls, 800 service calls, and automatic tie-trunk calls.

- Answer supervision

  The switch sends answer supervision to the central office when a call connects to an extension in the hunt group or an announcement. Charging for the call, if applicable, then begins.

Interactions

- Attendant Call Waiting

  Attendant Call Waiting does not work for calls that the attendant sends to a hunt group. It does work for calls to individual hunt-group members.

- Attendant Return Call

  Attendant Return Call does not work for calls that the attendant sends to a hunt group.

- Automatic Callback

  Automatic Callback does not work on calls to a hunt group.

- Automatic Call Distribution

  ACD does not work with circular station hunting.

- Call Detail Recording

  For each call, the system can record the associated hunt-group extension or member extension that answered.

- Internal Automatic Answer

  Internal calls to a hunt-group member are eligible for IAA.

- Leave Word Calling

- A hunt group can receive and store LWC messages. The following people can retrieve LWC messages:

  — One member of the hunt group

  — A covering user of the group

  — A system-wide message retriever

  The message retriever must have a phone display and proper authorization. If the message retriever is a member of the hunt group, you can assign to that member a remote Automatic Message Waiting lamp to indicate when the hunt group has an LWC message.

  Night Service — Hunt Group

  When Night Service is active for a hunt group and the night service destination is another hunt group, the caller hears the forced announcement of the first hunt group, if administered. The system then redirects the call to the night-service hunt group.

- Priority Calling

  The system treats a priority call to a hunt group the same as a nonpriority call, except that the extension receives a distinctive 3-burst ring.

- Queuing

  Queuing does not work with circular station hunting.
- Terminating Extension Group

  A Terminating Extension Group cannot be a member of a hunt group.
- Vectoring

  Call vectoring does not work with circular station hunting.

Related topics

Refer to ‘‘Announcements/Audio Sources’’ on page 588 to assign analog, DS1, auxiliary trunk, or integrated announcements, audio/music sources, or any desired combination of announcements and audio sources.

Refer to ‘‘Data modules’’ on page 679 for information about and field descriptions on the Announcement circuit pack.

Refer to ‘‘Coverage Path’’ on page 672 for information on automatic redirection of calls to answering positions.

Refer to ‘‘Trunk Group’’ on page 1167 for information about where incoming calls terminate or assigning an extension number to night service.

Refer to ‘‘Setting up hunt groups’’ on page 222 for information on how to set up hunt groups.

Incoming Call Line Identification

Your switch collects the calling party name and number (Incoming Call Line Identification, or ICLID) received from the central office (CO) on analog trunks.

Detailed description

Your switch stores and displays up to 15 characters of ICLID information received on analog trunks. If the information is longer than 15 characters, it truncates to 15 characters. If the caller ID information is not received, the trunk group name and trunk access code appear on the display.

Calling party information appears on all Avaya MultiVantage digital phones with 40-character or 32-character displays. In the US, the CO sends both calling party name and number, if they are available. In Japan, the CO sends only the calling party number.

Interactions

- Distributed Communications System (DCS)

  If the Avaya MultiVantage has both DCS and ISDN displays, the ICLID information displays in DCS formats.

Intercom

If you have users who call each other frequently, you can help them communicate more quickly. With the intercom feature, you can allow one user to call another user in a predefined group just by pressing a couple of buttons. You can even administer a button that always calls a predefined extension when pressed.

Brief description

You control which phones can make intercom calls to each other by putting them in groups called “intercom groups.” Once a set of phones have been added to the group, you allow users to make intercom calls by administering one or both of the following feature buttons on their phones:

Automatic Intercom	This button always calls one predefined phone in the same intercom group when pressed. You specify the destination extension for this button.
Dial Intercom	This button allows users in an intercom group to call anyone else in the same group. The caller lifts the handset, presses the Dial Intercom button, then dials a 1- or 2- digit code for the extension they want to reach.

Phones with one or both of these features can belong to the same group.

Intercom groups

- You can create up to 32 intercom groups on one Avaya DEFINITY Server.
- Each group can have up to 32 extensions in it.
- It’s OK to assign the same extension to different groups.
- Intercom calls are only possible between extensions in the same group.
- Any group member with a feature button for dial intercom can make an intercom call to any other member in the group.

Phones

- You can assign any type of phone to an intercom group. However, only multiappearance phones can make and receive intercom calls. Single-line phones can only receive intercom calls. Multiappearance phones must have at least one open or available call appearance to receive intercom calls.
- Phones receiving an intercom call make a unique alerting sound. If the phone has an intercom button with a status lamp, the lamp flashes.
- You can administer an automatic intercom connection between 2 phones even if their classes of restriction don't allow other calls between them.

Interactions

- Bridged Appearances

  Bridged appearances can't receive intercom calls.
- Call Coverage

  Intercom calls do not follow a coverage path unless the caller activates Go To Cover.
- Call Forwarding

  Intercom calls cannot be forwarded off-net.
- Call Pickup/Directed Call Pickup

  Intercom calls are not included in the call pickup alerting count.
- Data Privacy and Data Restriction

  Extensions with either of these features active cannot originate intercom calls.

Related topics

Refer to ''Using phones as intercoms'' on page 473 to administer intercom capabilities on phones.

Internal Automatic Answer

Internal Automatic Answer (IAA) provides convenient hands-free answering of internal calls to users on multifunction stations with a speakerphone or a headphone.

Detailed description

An eligible call can be answered automatically via IAA if the user activates IAA at the answering telephone and the user is able to accept an incoming call. A telephone is unable to accept an incoming call if it is off-hook, is in the process of dialing digits, or has a call on hold.

The following internal calls are eligible for IAA, depending on how you administer the Internal Auto-Answer of Attd-Extended/Transferred Calls field on the Feature-Related System Parameters screen:

- Station-to-station voice calls, with both telephones on the same switch (includes redirected intraswitch calls). Set the Internal Auto-Answer of Attd-Extended/Transferred Calls field to **transferred** or **both**.
- Internal call from another switch node in a Distributed Communications System (DCS) configuration when the origin of the call is known to be an internal, non-attendant telephone on that switch (includes redirected inter-DCS calls). Set the Internal Auto-Answer of Attd-Extended/Transferred Calls field to **transferred** or **both**.
- Attendant-extended external calls. Set the Internal Auto-Answer of Attd-Extended/Transferred Calls field to **attd-extended** or **both**.

The following calls are *not* eligible for IAA:

- Calls from public-network trunks (including personal central office line (PCOL))
- Calls from non-DCS tie trunks
- Automatic Callback calls
- Automatic Circuit Assurance calls
- Data calls
- Attendant-extended external calls if the Internal Auto-Answer for Attd Extended/Transferred Calls field is set to **transferred** or **none**
- Calls that the system redirects because of a queue overflow of Emergency Access to the Attendant calls
- Calls when the receiving station’s Active Station Ringing field is set to **continuous**

IAA Feature Operations

With IAA, you can assign a single programmable feature button (IAA) to telephones. When the user presses the IAA feature button, the button lamp lights and the system activates IAA. Pressing the same button again deactivates IAA and turns off the status lamp. (Pressing the feature button has no effect on a currently-active call or a ringing call.) The IAA button may be toggled on or off at any time, regardless of the state of the telephone. Using the speakerphone to place calls does not affect the state of IAA.

The calling telephone receives a tone when its call is answered automatically by a telephone with IAA. The called telephone receives a tone (a ring ping) and then goes off-hook when automatically answering an IAA-eligible call. The answering telephone's speaker and microphone are both turned on.

If a user has IAA active and is currently busy on a call or is in the process of dialing digits, subsequent incoming calls are treated as if IAA were not activated.

Considerations

- Users should always deactivate IAA when leaving the work area. Otherwise, incoming calls are unintentionally answered by the unattended station, and do not go to coverage.
- A 602A terminal is off-hook when the headset or speakerphone is connected. Therefore IAA answers a call if all other call appearances are idle.

Interactions

- Attendant Console

  IAA is not available with Attendant Console.
- Automatic Answer

  You cannot administer both IAA and Automatic Answer simultaneously on the same telephone.
- Automatic Call Distribution

  Calls directed to an ACD split are eligible for IAA.
- Automatic Callback

  Callback calls via Automatic Callback are not answered automatically by IAA.
- Automatic Circuit Assurance

  Calls generated by ACA are not eligible for IAA.

- Bridged Call Appearance — Multiappearance Telephone

  Calls terminating on a bridged call appearance are not eligible for IAA at the bridged station, even if the bridged station has IAA active. (IAA can be used by the principal station to answer the call.)

- Bridged Call Appearance — Single-Line Telephone

  Calls terminating to a bridged call appearance are not eligible for IAA at the bridged station, even if the bridged station has IAA active.

- Call Coverage

  If an internal call is redirected to another telephone by Call Coverage redirection criteria, then that call is eligible for IAA at the redirected telephone.

  IAA does not apply to calls to the original called extension when:

  — The called telephone has activated Send All Calls

  — The calling telephone has selected Go to Cover before placing the call

  Calls directed to a coverage answering group are not eligible for IAA.

  NOTE:
  If you set the coverage path for a station to All Calls and that station activates IAA, the first coverage point hears a ring, the principal station automatically answers, and the coverage-simulated bridge is dropped. The coverage station rings, but is not able to answer the call because the coverage-simulated bridge has been dropped.

- Call Forwarding

  Calls to a station with IAA and Call Forwarding active are forwarded and are not answered by the station dialed.

  NOTE:
  If the forwarded-to station is internal and has IAA active, it automatically answers the redirected call.

- Call Park

  If you are using Deluxe Paging and Call Park times out, the call returns to the originating station that parked the call and is eligible for IAA.

- Call Pickup

  Internal calls to a telephone in a Call Pickup group are eligible for IAA. If the called extension in a Call Pickup group has IAA-active, the call is automatically answered. A telephone with IAA active is not able to automatically answer calls to other telephones in its Call Pickup group.

- Conference

  Internal conference calls can be answered automatically via IAA. If more than one party has joined a conference call through automatic answer, the parties remain connected until they disconnect or the controlling party drops the call.

- Data Call Setup

  Data calls are not eligible for IAA.

- Direct Department Calling and Uniform Call Distribution

  Internal calls to a DDC or UCD group member are eligible for IAA.

- Distributed Communications System

  If a call is from an internal telephone on another switch in a DCS configuration, then that call is considered internal and is eligible for automatic answer.

- Do Not Disturb

  Do Not Disturb preempts IAA at the called telephone.

- Go to Cover

  IAA does not apply to calls to the original called extension when the calling telephone has selected Go to Cover before placing a call.

- ISDN-BRI

  IAA is not available with ISDN-BRI terminals.

- Loudspeaker Paging — Deluxe Paging

  When you are using Deluxe Paging and Call Park times out, the call returns to the originating station that parked the call and is eligible for IAA.

- Ringback Queuing

  Automatic calls generated by Ringback Queuing are not eligible for IAA.

- Send All Calls

  IAA does not apply to calls to extensions with Send All Calls is active.

- Terminating Extension Group

  Calls to a Terminating Extension Group extension are not eligible for IAA. However, calls placed to an individual extension are eligible.

ISDN service

The Integrated Services Digital Network (ISDN) provides a message-oriented signaling method that allows information to be sent along with a call and gives you access to a variety of public and private network services and facilities. The ISDN standard consists of layers 1, 2, and 3 of the Open System Interconnect (OSI) model. Avaya MultiVantage can be connected to an Integrated Services Digital Network using standard frame formats: the Basic Rate Interface (BRI) and the Primary Rate Interface (PRI).

An Integrated Services Digital Network provides end-to-end digital connections and uses a high-speed interface that provides service-independent access to switched services. Through internationally accepted standard interfaces, an Integrated Services Digital Network provides circuit or packet-switched connections within a network and can link to other ISDN-supported interfaces to provide national and international digital connections.

NOTE:
This manual does not contain procedures for working with ISDN trunk groups. Due to the complexity of ISDN technology and the potential consequences of errors, ask your Avaya representative to assist you in planning, installing, and administering ISDN trunks.

Brief description

ISDN supports the following:

- Call-by-Call Service Selection (CBC)
- Distributed Communications System (DCS). (Only ISDN-PRI supports DCS+ and DCS with Rerouting)
- Electronic Tandem Networks (ETN)
- Facility Associated Signaling (FAS) and Non-Facility Associated Signaling (NFAS) (Only ISDN-PRI supports this.)
- Generalized Route Selection (GRS)
- Call Identification Display — Calling Party Number (CPN) and Billing Number (BN)
- Administered Connections and Access Endpoints
- Interworking (a mixture of ISDN and non-ISDN trunks and stations)
- Wideband Switching (H0, H11, H12, and NxDS0 — only ISDN-PRI supports this.)
- QSIG Multivendor Connectivity

- Lookahead Inteflow
- Lookahead Routing
- Usage Allocation

Screens used to administer ISDN

Refer to Access Endpoint to administer Access Endpoints and Wideband Access endpoints.

Refer to ISDN trunk group for detailed information on the fields used to administer:

- — Supplementary Service Protocol (supports public network connection)
- — Calling Number (supports CPN)
- — Incoming Call Handling Treatment Table (supports digit manipulation, CPN or BN requests, and night service destinations)
- — CBC Trunk Group Usage Allocation screen (supports Call-by-Call Service Selection Usage Allocation Plans)
- — CBC Trunk Group Allocation Plan Assignment Schedule (supports Call-by-Call Service Selection Usage Allocation Plans)
- — Wideband Support Options (supports Wideband Switching)

Refer to ''ISDN Numbering — Private'' to administer private numbering plans.

Refer to ''ISDN Numbering — Public/ Unknown'' to administer ISDN call identification displays.

Refer to the DEFINTY services documentation for information about the following screens: ISDN TSC Gateway Channel Assignments, Network Facilities (supports usage allocation used in Call-by-Call Service Selection), and Signaling Group (used to define a group of B-channels for which a given D-channel or D-channel pair carries signaling information).

Transmission rate and protocols

In ISDN-PRI, the transmission standard for layer 1 (the physical layer) is either DS1 T1 or E1. The DS1 T1 (used in North America and Japan) is a digital-transmission standard that carries traffic at the rate of 1.544 Mbps, and the E1 (used in Europe) carries traffic at a rate of 2.048 Mbps. The “D” (data) channel multiplexes signaling messages for the “B” (bearer) channels carrying voice or data. In a T1, when a D-channel is present, it occupies Channel 24. In an E1, when a D-channel is present, it occupies channel 16.

Avaya MultiVantage offers several administrable protocols, each of which provides a different set of ISDN services. Refer to ‘‘DS1 Trunk Service’’ on page 1542 for more information. These protocols are discussed in detail later in this section. The following combination of services, including but not limited to Basic Call, Basic Supplementary Services, Supplementary Services with Rerouting, Display, and QSIG Networking are supported on the ISDN-PRI interface. Available services outside the United States vary from country to country.

With ISDN, Avaya MultiVantage interfaces with a wide range of other products including switches, network switches, and host computers. These products include earlier MultiVantage systems, public network switches (for example, 4ESS, 5ESS, and Northern Telecom DMS250), and other products that adhere to the ISDN signaling protocol.

As an example of how ISDN is used in private- and public-network configurations , see the following figures. For example, ISDN can be used to connect a switch to a public-switched network, to other switches, and to computers:

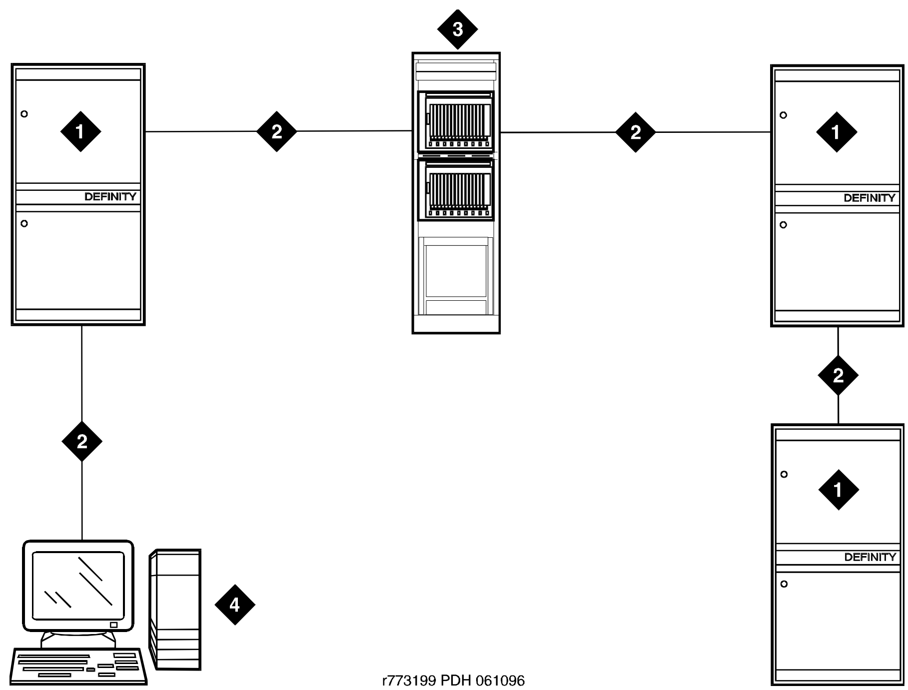


Figure Notes

1. Avaya Media Server
2. ISDN trunk
3. Public switched network
4. Host computer

Figure 59. ISDN network configuration

AT&T Switched Network Protocol

Avaya MultiVantage supports the AT&T Switched Network Protocol described in the TR41449 (for 4ESS to common carrier) and TR41459 (for 5ESS to CO) ISDN protocol standards as defined by AT&T. This protocol is used when the DS1 circuit pack is administered for Country Code 1, Protocol Version a. The AT&T Switched Network provides you with the following services.

Access to AT&T Switched Network Services

ISDN provides access to AT&T Switched Network Services. The definition of the Service Type field on the ISDN Trunk Group screen includes a table that outlines these switched-network services. An ISDN trunk group may be dedicated to a particular feature. Alternately, an ISDN call-by-call trunk group may provide access to several features. For a description of the services accessible via ISDN (either via dedicated or call-by-call trunk groups), refer to ''Call-by-Call Service Selection'' on page 1513.

Call Identification Display

ISDN Call Identification Display provides a transparent name and number display for all display-equipped telephones within an ISDN network. The feature is transparent in that the same information can be provided at all ISDN facilities. Telephones using this feature should be digital telephones with a 40-character alphanumeric display. The Merlin hybrid sets with 32-character displays (7315H and 7317H) also support this feature.

ISDN Call Identification Display is provided in addition to the normal Telephone Display and Attendant Display features when the network supports end-to-end ISDN connectivity. When both ISDN and DCS display information are received, the switch can display either the DCS or ISDN call identification information. If only ISDN display information is received, information displays in ISDN format.

The display fields that may be used for ISDN are: Name, Number, Miscellaneous Call Identification, and Reason for Call Redirection. The display information varies, depending on the type of call, how the call is handled (for example, whether it is redirected or not), and the information is available on the call.

CPN/BN to Host Call Identification

The CPN/BN to Host Call Identification enables CPN and BN information to be passed from the switch to the ISDN Gateway, so that the ISDN Gateway can forward the information to a host for data-screen delivery to agents in an ACD split.

By delivering call-identification information such as CPN/BN and switch information such as the answering-agent's extension to an adjunct network (ISDN Gateway), the adjunct automatically delivers data screens to agents for new calls and call transfers.

The figure below shows a simplified diagram of a CPN- and BN-to-host arrangement. The ISDN Gateway is a UNIX or MSDOS computer connected to the switch on one side and to a host computer on the other side. The connection to the switch is over a synchronous interface with BX.25 protocol.

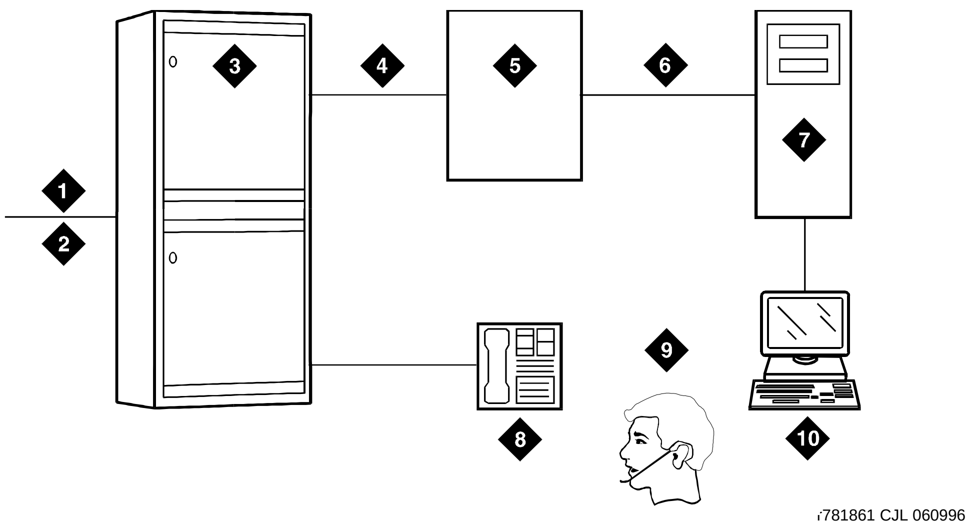


Figure Notes

1. ISDN trunk
2. SID/ANI
3. Avaya Media Server
4. BX.25
5. ISDN Gateway
6. Existing interface
7. Host computer
8. Telephone
9. ACD agent position
10. Data terminal

Figure 60. CPN- and BN-to-host configuration

Private network services

In addition to providing access to switched-public networks, ISDN provides private-network services by connecting Avaya MultiVantage in an Electronic Tandem Network (ETN), Distributed Communications System (DCS), or QSIG Network. This gives you more efficient private networks that support new integrated voice and data services. ETN, DCS, and QSIG networking services are provided as follows.

ETN services

Avaya Media Servers that function as tandem nodes in an ETN can be interconnected using DS1 trunking facilities with ISDN. All signaling between the tandem switches is done with ISDN D-channel and normal ISDN protocol. The ISDN can also be used to connect ETN tandem and main switches. In this case, the main switch collects all of the address digits from local users as well as users at other satellite and tributary switches, and originates a call over ISDN to the tandem switch.

Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) are used with ISDN and DS1 trunking facilities to access ETN facilities. AAR and ARS are used to collect the dialing information for the call that is originated from the main switch.

DCS services

ISDN-PRI facilities can be used in a DCS arrangement whenever tie trunks are used to connect the DCS nodes. Most DCS features are not affected by ISDN-PRI. However, there is a minor impact on a few of the DCS features, as far as the functions that the local and remote switches perform.

QSIG services

QSIG networking provides compliance to the International Organization for Standardization (ISO) ISDN private-networking specifications. The QSIG Networking platform is supported over the ISDN Basic Call setup protocol. Avaya MultiVantage supports QSIG Supplementary Services.

Wideband Switching (ISDN-PRI only)

Wideband Switching provides support for services that require large bandwidth, such as high-speed video conferencing. Wideband also supports multiple channel calls end-to-end. These services have traditionally been handled by dedicated facilities. With Wideband Switching, dedicated facilities are no longer a requirement for these large bandwidth services.

Call-by-Call Service Selection

Call-by-Call Service Selection allows the same ISDN trunk group to carry calls to a variety of services or facilities. Embodied in this feature is the ability to allocate usage. It provides significant flexibility for creating user-defined incoming and outgoing services and is used on any ISDN trunk group.

Access to Software Defined Data Network

With ISDN, the SDDN service may be accessed. SDDN provides virtual private-line connectivity via the switched public network. The services provided by SDDN include voice, data, and video applications. SDDN services complement the ISDN voice services.

Access to Switched Digital International

Switched Digital International (SDI) provides 64 kbps of unrestricted connectivity to international locations via the AT&T Switched Network. It is also the backbone for the AT&T International ISDN network. SDI complements the ACCUNET digital service already available to United States locations. This service can be accessed using Call-by-Call Service Selection. SDI provides economical high-speed data transfer to international locations.

National ISDN-2 Services

Avaya MultiVantage supports National ISDN-2 (NI-2), which offers many of the same services as the AT&T Switched Network protocol. The NI-2 protocol is used when the DS1 circuit pack is administered for Country Code 1, Protocol Version b.

NI-2 provides users with the following services:

- Calling Line Identification
- Non-Facility Associated Signaling (ISDN-PRI only)
- D-Channel Backup (ISDN-PRI only)
- Wideband Switching (ISDN-PRI only)
- Call-by-Call Service Selection

Calling Line Identification

Calling Line Identification for NI-2 is essentially Calling Party Number (CPN) identification, as previously described.

Non-Facility Associated Signaling (ISDN-PRI only)

Non-Facility Associated Signaling (NFAS) allows an ISDN-PRI T1 or E1 Interface D-channel (signaling channel) to convey signaling information for B-channels (voice and data) on ISDN-PRI T1 or E1 facilities other than the one containing the D-channel. See ‘‘Facility and Non-Facility Associated Signaling’’ on page 1558 for more information.

D-Channel Backup (ISDN-PRI only)

D-Channel Backup is provided to improve reliability in the event of a signaling-link failure. See ''Facility and Non-Facility Associated Signaling'' on page 1558 for more information.

Wideband Switching (ISDN-PRI only)

Wideband Switching for NI-2 is essentially the same as that of the AT&T Switched Network ISDN-PRI protocol.

Call-by-Call Service Selection

Call-by-Call Service Selection for NI-2 is essentially the same as that for the AT&T Switched Network ISDN-PRI protocol.

ISDN interworking

ISDN interworking allows calls to use a combination of both ISDN and non-ISDN trunking and station facilities. A non-ISDN trunking facility is any trunk facility supported by the system that does not use the ITU-T recommended Q.931 message set for signaling. Non-ISDN trunking facilities include facilities such as analog trunks, AVD DS1 trunks, and DS1 trunks with bit-oriented signaling (robbed-bit or common channel).

Avaya MultiVantagesupports the conversion of ISDN signaling to non-ISDN in-band signaling and the conversion of non-ISDN in-band signaling to ISDN signaling for interworking purposes.

A mixture of ISDN and non-ISDN signaling is required in order to provide end-to-end signaling when using different types of trunk or station facilities on a call. See the figure below for an example of interworking.

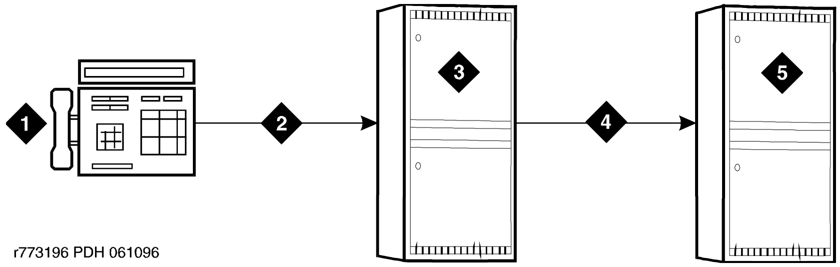


Figure Notes

1. Call from network to Switch B
2. ISDN trunk
3. Switch A
4. Non-ISDN trunk
5. Switch B

Figure 61. ISDN and non-ISDN interworking

In this example, a call for someone at Switch B comes into Switch A. Interworking allows the ISDN signaling of the call to be converted at Switch A to non-ISDN in-band signaling before the call forwards to Switch B. Even though the call comes into Switch A on an ISDN trunk, Switch A can send the call to Switch B over a non-ISDN trunk by converting the signaling information.

The system provides accurate CDR billing information on calls that are not interworked. Accuracy of CDR billing information on interworked calls is equivalent to the accuracy provided by the public network.

Avaya MultiVantage supports sending a non-ISDN trunk name as the connected name. Therefore, a non-ISDN trunk name can be sent as the connected name even when a call starts out as an ISDN call but is interworked over non-ISDN trunks.

Call Identification Display

Two types of identification numbers are provided with ISDN and may be used in the various types of displays used with ISDN. The two types of identification numbers are as follows:

- Calling Party Number (CPN): A 0–15 digit DDD number associated with a specific station. When a system user makes a call that uses ISDN, that user's CPN is provided by the system for ISDN. ISDN public-unknown numbering or ISDN private numbering screens are administered to create a 0–15 digit CPN from a local station number.
- Billing Number (BN): The calling party's billing number, which is provided to an inter-exchange network via Equal Access or Centralized Automatic Message Accounting (CAMA). This number is stored at either a local or network switch. If a customer is connected directly to the AT&T Switched Network, the BN is the customer's billing number stored in that network. If the CPN is not provided on an incoming ISDN call, the network uses the BN for the station identification number.

The following types of display information are provided with ISDN:

- Calling Party's Number

  The calling party's number appears on the called party's display. This number is provided only if the outgoing ISDN trunk group is administered to send the CPN, and if ISDN public-unknown numbering or ISDN private numbering screens are administered to create a CPN. On calls incoming to a system, the network may provide either the CPN or BN as the calling party's number. Extensions and 12-digit international numbers display without dashes. Dashes are only used for 7-digit and 10-digit numbers when North American Area Code is enabled on the Dial Plan screen.

- Calling Party's Name

  The calling party's name appears on the called party's display. On calls generated from an Avaya DEFINITY Server, the caller's name is provided if the ISDN trunk group is administered to send the name to the network. On calls incoming to an Avaya DEFINITY Server, the (public or private) network may provide the caller's name. If the caller's name is not available, the called party's display shows "CALL FROM" instead, followed by the calling party's number (if available).

- Connected Party's Number

  The connected party's number appears on the caller's display. On calls generated from an Avaya DEFINITY Server, callers' displays may show the digits dialed as the call is made. If the (public or private) ISDN network provides the connected party's number, the calling party's display is updated to show the connected party's number. The format of the connected party's number is the same as that of the calling party's number described previously on calls incoming to an Avaya DEFINITY Server. The 0–15 digit number of the party who answers the call is provided to the ISDN network only if the incoming ISDN trunk group is administered to send connected number to the network and ISDN public-unknown numbering or ISDN private numbering screens are administered to create a CPN.

  NOTE:
  The connected party may be the party actually called, in the event the call is transferred before the connected party answers the call.

- Connected Party's Name

  The connected party's name appears on the calling party's display. On calls generated from an Avaya DEFINITY Server, the (public or private) ISDN network may provide the connected party's name to the Avaya MultiVantage, when the call is answered. If the connected party's name is not available, the calling party's display shows *ANSWERED BY*, followed by the connected party's number (if available).

  On calls incoming to an Avaya DEFINITY Server, the connected party's name is provided if the incoming ISDN trunk group is administered to send the name to the network.

Depending on how the switches involved in a call are configured, parties may see none, some, or all the information described above.

Displays for redirected calls

Features such as Call Coverage, Call Forwarding All Calls, Bridged Call Appearance, or Call Pickup redirect calls from the called party's extension to some other destination. Once the redirected call has been connected at its new destination, the displays for the calling, called, and connected parties are as follows:

— Calling Party Display

a= CONNECTED NAME   CONNECTED NUM   MISCID

— Called Party Display

This is the display of the party the caller originally dialed. If this party bridges onto the redirected call after it has been answered, they see:

a= CONFERENCE 2

In this situation, the connected party's display (see below) shows the same information. The calling party's display is also updated if the calling and called parties are on the same switch.

— Connected Party Display

The connected party is the party who answers the redirected call.

a= CALLING ID   to   CALLED ID   R

The R indicates the reason for redirection. The CALLING ID and the CALLED ID may be the name or the number, depending on the information received from the far end.

Displays for conference calls

Both terminal and attendant conference calls are identified as calls with "n" number of conferees. This display information generates locally and does not change the display on another switch. If the conference call eventually drops back to a two-party call, the original display information is restored. However, when two DCS and/or ISDN calls (or any possible combination of each) are conferenced and revert to a two-party call, the trunk group of the remaining call displays.

Displays for calls to hunt groups

On ISDN calls to a hunt group extension, the caller's display identifies either the name of the hunt group or the name of the group member who answers the call, depending on hunt group administration.

Displays for calls to Terminating Extension Groups (TEG)

On ISDN calls to a TEG, the caller's display identifies either the group or the group member who answers the call, depending on administration.

Caller Information Forwarding

With CINFO you can use a vector *collect digits* step to retrieve caller entered digits (ced) and customer database-provided digits (cdpd) supplied by the network in an incoming call's ISDN SET UP message. ISDN is required if the CINFO comes from the network.

Facility Restriction Level and Traveling Class Mark

The TCM used to pass on the originating facility's FRL is sent by ISDN facilities in the SETUP message only if the trunk services type is tandem.

Information Indicator Digits (II-digits)

With II-digits you can make vector-routing decisions based on the type of the originating line. II-digits are provided for an incoming call by ISDN-PRI. It is a generally available ISDN AT&T Network service.

Malicious Call Trace (MCT)

ISDN calling number identification is sent when MCT notification is activated on an ISDN trunk.

Multiple Subscriber Number (MSN) - Limited

The ISDN standard MSN feature lets you assign multiple extensions to a single BRI endpoint. A side effect of supporting the NT interface is the MSN feature works with BRI endpoints allowing the Channel ID IE to be encoded as "preferred." The endpoint must be administered as the far end of an NT-side ISDN-BRI trunk group. Also, you must use the Uniform Dial Plan (UDP) feature to assign the desired extensions to the "node" at the far end of the trunk group.

Overlap Sending

You can administer overlap sending on AAR and ARS calls routed over ISDN trunk groups. This allows you to send and receive digits one digit at a time instead of enbloc. (With enbloc, digits are not sent until the entire group of digits is received).

Interworking between TGU/TGE trunks and ISDN (Italy)

This feature modifies ISDN messaging operations in systems that use TGE/TGU trunks to network satellite switches. Messaging from the Avaya MultiVantage provides appropriate ringback or busy tone to the calling party.

Interactions

- Australia Malicious Call Trace

  Avaya MultiVantage with a BRI connection to the Australian public network is able to notify the network if a user in a private network invokes the Malicious Call Trace feature. This works on Australian national connections.

- Direct Inward Dialing

  Some public network operators may not offer full Direct Inward Dialing service on BRI trunks, but instead may offer the BRI equivalent, which typically is called Multiple Subscriber Number (MSN). This is the case if the public network treats the BRI as an endpoint interface rather than a trunk group interface.

  In such a case, the network only routes up to 10 public numbers to a particular pair of BRI trunks. Furthermore, the network may not let calls overflow from one BRI trunk to another.

- D-Channel Backup

  D-Channel Backup is not supported on BRI connections.

- Distributed Communications System

  If both DCS and ISDN features are provided over the same facility with an Avaya DEFINITY Server, DCS displays generally override ISDN displays. However, with Avaya MultiVantage, the ISDN connected name and number can override the DCS called name and number if the Display Connected Name/Number for ISDN DCS Calls field is y on the Feature-Related System Parameters screen.

  BRI trunks support DCS if using a BX.25 link to transport the DCS messages. DCS+, also known as DCS Over ISDN D-Channel, according to the AT&T protocol, is not supported on BRI trunks.

- Facility Test Calls

  Neither BRI or PRI trunks support Facility Test Calls.

- France VN4 Protocol

  The France national VN4 protocol is supported on BRI trunks as ETSI.

- Generalized Route Selection

  BRI trunks are capable of carrying 56Kbps or 64Kbps data calls. The link coding that restricts certain PRI trunks to 56Kbps only does not apply to BRI trunks.

- German 1TR6 Protocol

  The German national 1TR6 protocol is not supported over BRI trunks.

- Message Sequence Tracer

  ISDN-BRI trunks support Message Sequence Tracer. However, certain filtering capabilities available for PRI trunks are not available. Specifically, it is not possible to filter BRI trunk messages based on incoming/outgoing calling/called number.

- Network Access - Public (LEC/AT&T/Other Carriers)

  Public-network access using BRI trunks is available but only in those countries that support point-to-point BRI connections. In the U.S., BRI access is offered only by the Local Exchange Carriers and not by Interexchange Carriers such as AT&T.

- Network Access - Private Premises Based

  Full support for private-network connections using BRI trunks is available.

- Non-Facility Associated Signaling

  Non-Facility Associated Signaling is not supported on BRI connections.

- Temporary Signaling Connections

  Avaya MultiVantage does not support Temporary Signaling Connections according to the AT&T protocol on BRI trunk interfaces. Only the QSIG NCA TSC protocol is supported on these interfaces

- Wideband Switching (NxDS0)

  Avaya MultiVantage does not support wideband switching on BRI connections.

Related topics

Refer to Administered Connections ‘‘Administered Connections’’ on page 1338 for detailed information on this feature.

Refer to ‘‘Call-by-Call Service Selection’’ on page 1513 for detailed information on this feature.

Refer to *Administration for Network Connectivity for Avaya MultiVantage™ Software* for information on Distributed Communications System (DCS) networks and on QSIG and QSIG features such as Call Diversion, Call Transfer, Name Identification, and Path Replacement.

Refer to ‘‘Facility and Non-Facility Associated Signaling’’ on page 1558 for information on assigning data channels to bearer channels on one or more circuit packs.

Refer to ''Generalized route selection'' on page 1567 for information on this feature that matches calls to appropriate transmission facilities based on the endpoints and the type of call (for example, voice or data).

Refer to ''Look-Ahead routing'' on page 1630 for information on balancing call volume across a network by automatically rerouting ISDN calls to other switches in the network.

Refer to ''Wideband Switching'' on page 1811 for information on providing high-speed end-to-end connectivity for applications (such as video conferencing) that require high bandwidth.

Leave Word Calling

Leave Word Calling (LWC) allows internal system users to leave a short preprogrammed message for other internal users. When the message is stored, the Automatic Message Waiting lamp on the called telephone lights. Users can retrieve LWC messages using a telephone display, Voice Messaging Retrieval, or AUDIX. Messages may be retrieved in English, French, Italian, Spanish, or a user-defined language.

Avaya MultiVantage also provides voice synthesis (either English or Italian) for LWC, depending on which voice-synthesis circuit pack is installed in the system.

The system can indicate that one telephone received a LWC message on a second telephone. The system lights a remote Automatic Message Waiting lamp at the remote telephone and the Automatic Message Waiting lamp lights at the called telephone. The Remote Automatic Message Waiting lamp is a status lamp associated with a button assigned for this purpose. Thus, an assistant's telephone could light when an executive receives a LWC message. If the executive calls to retrieve messages, the assistant knows at a glance if any messages have been left.

Users without telephone Display can have their messages retrieved by a system-wide message retriever or by covering users in their Call Coverage path. They can also use Voice Message Retrieval.

The system restricts unauthorized users from displaying, canceling, or deleting messages. The Lock function restricts a telephone and the Unlock function releases the restriction. Users activate Lock by dialing a system-wide access code. They cancel Lock by first dialing a system-wide access code and then an Unlock security code unique to the telephone. These functions apply only to the telephone where the function is active. You can assign a status lamp to show the lock status of the telephone.

Leave Word Calling Log External Calls allows the switch to monitor when an external call is not answered. The switch keeps a record of up to 15 calls, provided information on the caller identification is available, and the phone's message lamp lights. The phone set displays the names and numbers of unsuccessful callers.

Considerations

- You can administer up to 10 telephones (or nine telephones and the attendant console group) as system-wide message retrievers.
- If the stored-message level reaches 95 percent of capacity, the status lamps associated with all Coverage Message Retrieval buttons in the system flash. These lamps continue to flash until the stored-message level falls below 85 percent. Authorized retrievers can selectively delete messages to gain storage space. Old messages are not purged automatically by the system.
- LWC messages cannot be stored, canceled, or retrieved for Vector Directory Number extensions.

Interactions

- AUDIX Interface

  LWC Cancel cannot be used to cancel an AUDIX message.
- Bridged Call Appearance

  A LWC message left by a user on a bridged call appearance leaves a message for the called party to call the primary extension for the bridged call appearance. When a user calls a primary extension and activates LWC, the message is left for the primary extension, even if the call was answered at a bridged call appearance.
- Call Coverage

  You can use LWC with or without Call Coverage. However, the two features complement each other. The Coverage Callback option of Call Coverage is provided by LWC. Also, a caller can activate LWC for the called party even if the call has been answered by a covering user.
- Centralized Attendant Service

  LWC Message Retrieval does not work with CAS.

- Conference

  A member of a conference call cannot activate LWC because the user cannot be uniquely identified. After LWC has been activated for a party on a conference or transfer, the conference or transfer originator cannot press Conference/Transfer a second time to return to the original call. The originator must select the call appearance button to return to the previously-held call.

- Expert Agent Selection

  When an EAS agent is logged into a telephone, the agent can only retrieve LWC messages left for that agent's login ID. To retrieve LWC messages left for that telephone, the agent must log out.

  When an EAS agent is logged into a telephone, its Message lamp defaults to tracking the status of LWC messages waiting for the telephone. However, you can assign the Message lamp to track the status of LWC messages waiting for the agent's login ID.

- Vector Directory Number

  LWC messages cannot be stored, cancelled, or retrieved through VDN.

Line Lockout

Line Lockout removes single-line extensions from service when users do not hang up after receiving dial tone or intercept tone for an administered length of time.

You can administer the system to play a special "howler" tone before locking an analog extension by setting the Station Tone Forward Disconnect field to **busy** on the ''Feature-Related System Parameters'' screen. For the howler tone to play, an Avaya representative must also enable the Howler After Busy field on the ''System Parameters Country-Options'' screen.

If you want the system to disconnect and optionally lock out users who have let an outgoing public network trunk call ring for an extended period of time, you can have Avaya enable the Disconnect on No Answer by Call Type field, also on the ''System Parameters Country-Options'' screen.

Line Lockout occurs when:

- A user does not hang up after the other party on a call is disconnected.

  The user receives the dial tone for 10 seconds and then receives the intercept tone for the length of time administered in Line Intercept Tone Timer on the Feature-Related System Parameters screen. If the handset remains off-hook, the telephone is taken out of service.

- A user pauses for 10 seconds between digits while dialing.

  The user receives intercept tone for 30 seconds. If the handset remains off-hook, the telephone is taken out of service.

The out-of-service condition remains in effect until the user hangs up.

Considerations

- The out-of-service condition provided by Line Lockout does not tie up switching facilities.
- Line Lockout does not apply to multiappearance telephones.

Related topics

See ‘‘Feature-Related System Parameters’’ on page 766, Time Before Off-hook Alert for more information.

Listed Directory Numbers

Listed Directory Numbers (LDN) has two capabilities that allow outside callers to access your attendant group, depending on the type of trunk used for the incoming call. You use one capability to allow attendant group access via incoming direct inward dial (DID) trunks. You use another capability to allow attendant group access via incoming central office (CO) and foreign exchange (FX) trunks.

Brief description

The system routes both incoming DID calls and incoming FX and CO calls to an attendant group, depending on how you administer the trunks.

How the system routes incoming DID trunk calls to the attendant group

Incoming DID calls route only to an extension. The LDN feature allows you to assign one or more extensions to an attendant group. The system uses the LDN extension, or extensions, to route calls to an attendant group.

How the system routes incoming FX and CO trunk calls to the attendant group

Incoming FX and CO trunks can terminate at an attendant group, although you administer your system to terminate the calls elsewhere. You can administer the system to terminate an incoming FX or CO trunk to one of the following:

- Attendant group
- Extension (This could be a VDN, an ACD split, a DDC group, a UCD group, a remote access extension, or any system extension.)

If you decide to terminate the call at the attendant group, *the system treats the call as an LDN call*.

Considerations

The number of listed directory numbers that you can assign depends on your system's configuration.

Interactions

- Night Service

  If you activate night-service capability and a night console is not assigned or is not operational, incoming LDN calls route as follows:

  — DID LDN calls route to a designated DID LDN night extension. If no DID LDN night extension is designated, DID LDN calls route to the attendant.

  — Incoming CO or FX trunk calls route to the night destination specified for the trunk group. If no night destination is specified for the trunk group, the calls route to the normal incoming destination for that trunk group.

  — Internal calls and coverage calls to the attendant route to the DID LDN night extension during night service.

Location by region

Location by Region provides a way to administer location by IP network region. This allows for the correct date and time information and trunk routing based on the IP network region.

Detailed description

Location by Region allows you the ability to have an IP phone registered with an Avaya DEFINITY Server, but located at a site with C-Hawk or Remote Office endpoints, to display the correct time and date worldwide. The IP phone can be administered in a different network region from other Avaya MultiVantage endpoints, and in the same “location” as the C-Hawk or Remote Office users. It allows IP endpoint users the ability to move from location to location and always have correct display information. Remote users are identified in a network region and location that routes them to correct 911 services or notifies them via announcements, that they are in a different 911 jurisdiction than the DEFINITY they are registered to.

Alerts/Interactions

911 Call Handling

Location by Region attempts to overcome a limitation in the emergency response system - USA Public Safety Answering Points (PSAPs) currently have no good way to contact each other. Even if you are able to speak, and you explain to a PSAP in one part of the country that you need help in another part of the country, the 911 staff that you are talking to have no fast way to relay your comments to a PSAP near where you are located. If you describe your location as out-of-state, you likely will to be told to hang up and find a local phone.

Earlier 911 call handling features were unable to handle IP telephones because there was no way to pop up a window on an IP telephone to let the IP telephone user know why their 911 call was blocked and advise them on the right thing to do. Now, you can choose to dedicate one location to handle such roaming IP telephones. That special location could have corresponding ARS routing tables that route all 911 calls to a repeating announcement saying something like “You are too far away from the switch for the [name of the home location]’s public safety office to be able to help you. Please call 911 from a local circuit switched phone.”

Look-Ahead routing

Look-Ahead Routing (LAR) allows the switch to reroute an outgoing ISDN call that is not completing. The switch attempts to complete the call along a different routing preference, or it tries the current routing preference a second time. If the current preference fails twice, the next routing preference is tried.

LAR can be administered at an origination switch or a tandem switch. It can be turned off at different points in the network to reduce network load. You use LAR with AAR and ARS, GRS, UDP. You can also use it with a Feature Access Code ISDN Access Code. A LAR field is administered on the Route Pattern screen.

NOTE:
When LAR is used in a mixed network of Avaya MultiVantage Systems and DEFINITY® ECS switches, LAR ends at the DEFINITY® ECS switch and calls are rejected the normal way. However, if a LAR-triggering cause value is passed back in the network to an Avaya DEFINITY Server that is enabled for LAR, LAR is attempted from that switch again.

Detailed description

LAR can be administered for each ISDN route preference in each Pattern Number. The maximum number of LAR attempts per call per switch is 2 times the number of route preferences in the route pattern. LAR can be administered at each intermediate node that the call may be tandemed through to allow the attempt of all possible routes.

You can control LAR by:

- administering it on a per route-preference basis
- partitioning trunks
- limiting the number of hop counts

LAR activation

LAR is active when a call is rejected with a cause value in the range of #34–#47 and #3 (no route to destination). The range of #34–#47 indicates congestion and that resources are unavailable. The following cause values activate LAR:

Cause Value	Cause Description
3	no route to destination
34	no circuit/channel available
38	network out of order
41	temporary failure
42	switching equipment congestion
43	access information discarded
44	requested circuit or channel not available
47	resources unavailable

NOTE:
When country code 13, protocol version a, is administered on the DS1 Circuit Pack screen, only the cause values #10 and #89 activate LAR.

LAR terminates when:

- call is successfully routed
- call is rejected with a non-LAR-triggering cause value
- no further route preference can be used to route the call

LAR measurement

You can measure the number of attempted and successful LAR reroutes. The Measurements LAR Route Pattern screen displays LAR measurements for a particular route pattern. See *Avaya MultiVantage System Monitoring and Reporting* for more information.

Interactions

- Automatic Circuit Assurance

  LAR rerouting attempts are recorded as short holding time calls.

- Distributed Communications Systems

  If a non-DCS trunk preference is selected for rerouting a DCS call, DCS feature transparency is lost. If LAR for a DCS call is done within the same DCS trunk group, feature transparency is not lost.

- QSIG Networks

  If a non-Supplementary Services B (SSB) trunk preference is selected for rerouting a SSB call, QSIG feature transparency is lost. If LAR for a QSIG call is done through either the same or another SS B trunk group, feature transparency is not lost.

- Ringback Queuing

  When a call originates and queues at the trunk group queue, the call can be placed in queue multiple times if LAR is active. The call originator can be called back each time the call is continued automatically.

- Satellite Hop Limit

  Satellite Hop Limit always takes precedence over LAR. When the maximum hop limit is reached for a route preference, the last call routing attempt is denied and the call is rejected with a cause value of #28 — invalid number format. This value does not activate LAR.

- System Measurements

  System resource use during LAR attempts are included in existing system measurements and performance reports. For more information about LAR system measurements, refer to *Reports for Avaya MultiVantage Software*.

Related Topics

Refer to Route Pattern for information on the Route Pattern screen.

Loss Plans

Loss plans determine the amount of loss (quieter) or gain (louder) applied on calls. Usually, your system uses a pre-defined loss plan that is based on the administered country code. In some circumstances, you may be able to change the loss plan used by your system on a per trunk or per phone basis.

CAUTION:

The values in the loss plan can significantly affect the quality of service that your users experience. Therefore, in order to change the loss plan you must thoroughly understand loss plans and your particular configuration. We recommend that you seek technical assistance from Avaya before making any modifications to the loss plan.

In order to be able to modify your loss plan, the Digital Loss Plan Modification field must be set to **y** on the System-Parameters Customer-Options screen, and the Customize field must also be **y** on the System Parameters Country-Options screen. An Avaya representative must enable these fields for you.

The 2 Party Loss Plan page of the System Parameters Country-Options screen allows you to set the gain or loss levels (in dB) between two parties on a call. Each row on this screen is considered a different loss group. You can assign a loss group to a particular phone or trunk by administering a value for the Loss Group fields on the Station and Trunk Group screens. This allows you to use different loss plans for different types of phones or different trunk groups.

The Tone and Conference Loss Plans page of the System Parameters Country-Options screen allows you to set the gain or loss levels (in dB) between all parties on a conference call, as well as the total gain or loss in a conference based on the number of parties.

IP endpoints connected via hairpinning or direct IP–IP are not under the control of the administrable loss plan.

Loudspeaker paging

You can connect Avaya MultiVantage to loudspeaker systems and allow users to page from their phones. You can administer up to 9 separate zones (sets of loudspeakers) on Avaya MultiVantage, so an announcement can be made to one group or location without disturbing people who don’t need to hear the announcement. Auxiliary trunks connect the speakers in each zone to ports on an auxiliary trunk circuit pack.

Brief description

Avaya MultiVantage offers 2 types of loudspeaker paging. You can use each separately, and you can also use both together.

Voice paging	Voice paging allows users to make announcements over a loudspeaker system from their phones. You can integrate voice paging and Call Park by enabling deluxe paging.
Chime paging	If frequent voice pages are undesirable, you can assign a unique series of chimes (a *chime code*) to each extension. The chime code assigned to that extension plays over the speakers whenever that extension is paged. Chime paging is sometimes called Code Calling Access.

How users place voice pages

With standard voice paging, users page by dialing the trunk access code assigned to the zone they wish to page. If users have an active call, they must manually put the call on hold or park it before they dial the trunk access code.

When deluxe paging is enabled, users can automatically park an active call when they page, as described below.

Users with multi-appearance phones

The following description only applies to systems with deluxe paging. To page and park an active call simultaneously, users with a multi-appearance phone press Transfer, dial the trunk access code + an extension number where the call will be parked, make the announcement, and press Transfer again. The paged party dials the answer back feature access code + the extension number and is connected directly to the parked call. If the paging user ends the page by pressing Conference instead of Transfer, they are conferenced with the parked caller and both are connected in a 3-way conference with the paged user when he or she responds. This is called “Meet-Me Conferencing.”

If the paging user doesn’t want to park the active call, deluxe paging also allows “Meet-Me Paging.” Paging users can put an active call on hold and make their page, announcing their own extension. When the paged party calls, the paging user can conference the call on hold or transfer it to the paged party.

Users with single-line phones

This description only applies to systems with deluxe paging. To page and park an active call simultaneously, users with a single-line phone press Recall or flash the switch hook, dial the trunk access code + an extension where the call will be parked, and press Recall again. The paging user is conferenced with the parked caller and both are connected in a 3-way conference with the paged user when he or she responds. In other words, Meet-Me conferencing is standard operation for users with single-line phones. The paged party dials the answer back feature access code + the extension number and is connected directly to the parked call.

If the paging user does not press Recall until the loudspeaker paging time-out interval expires, they hear confirmation tone and the active call is automatically parked on their extension. When the paged party answers the call, they are connected to the paging party who can then transfer the call to the calling party.

How users place chime pages

Users page by dialing the trunk access code for a zone followed by the extension of the person they want to page. The system matches the extension dialed to its assigned code and plays the code over the loudspeakers. If users have an active call when they start to page, the call is automatically parked on the extension dialed in the page. Paged parties may retrieve the parked call normally.

Auxiliary paging systems

Avaya MultiVantage requires a separate port for each paging zone and supports a maximum of 9 zones. If you have more than 9 zones or don’t want to allot that many ports for paging, Avaya can provide auxiliary paging systems. These systems can support many zones from 1 port. They can also provide additional capabilities such as two-way communication through the loudspeaker system (the person paged can speak directly to the pager over the loudspeaker).

For more information, contact your Avaya representative.

Restrictions on loudspeaker paging

These restrictions apply to both voice, deluxe voice, and chime paging:

- A paging call can't be placed on hold, included in a conference call, or transferred. Ringback queuing doesn't work with loudspeaker paging calls either.
- Users with any of the following restrictions cannot page:
  - Controlled restriction
  - Manual originating line service
  - Origination restriction
  - Miscellaneous trunk restriction
- A user with a single-line phone will not hear a call-waiting tone if they get a call while they're paging.
- Listed directory number and direct inward dialing calls cannot access the paging system. However, attendants can park incoming calls and page.
- Remote users (such as remote access users and tie-trunk users) who are paging cannot use # to park calls on their own extensions.

Interactions

- Bridged Call Appearance

  If a parked call includes a shared terminating extension group, a shared PCOL, or a redirected call with a temporary bridged appearance, the maximum number of off-hook parties on the call is five, instead of six. The sixth position is reserved for the answer-back call.

- Call Coverage

  If a coverage call is parked by deluxe paging, the temporary bridged appearance at the principal extension is maintained as long as the covering user remains off-hook or places the call on hold.

- Call Park

  If a call is parked by deluxe paging and the time-out interval expires, the call normally returns to the paging user. However, with remote access and tie trunk access, the call returns to the attendant. If unanswered, the call follows the coverage path of the paging user.

- Call Pickup

  If you use call pickup or directed call pickup to answer a call and then park it by deluxe paging, a temporary bridged appearance at the principal extension is maintained if you remain off-hook or place the call on hold.

- Conference — Attendant and Terminal

  Paging calls cannot be conferenced.

- Data Call Setup

  If the Data button has been pressed for modem pooling, access to paging is denied.

- Data Privacy

  If a call has Data Privacy activated and you park it by deluxe paging, Data Privacy for that call is automatically deactivated.

- Hunt Groups

  If a hunt-group member parks a call using deluxe paging, the call is parked on the member's own extension, not the hunt-group extension. You cannot park calls on a group extension by dialing the extension as a call-park destination.

- Night Service

  If a night-station user parks a Night Service call with deluxe paging, the call is parked on the night station's primary extension.

- Personal Central Office Line

  If a PCOL call is parked by deluxe paging, the temporary bridged appearance of the call is maintained at the PCOL extension until the call is disconnected.

- Terminating Extension Group

  If a TEG member parks a call using deluxe paging, the call is parked on the member's extension, not the group extension. You cannot park calls on a group extension by dialing the extension as a call-park destination.

- Transfer

  Paging calls can't be transferred.

Chime paging

- Abbreviated Dialing

  Don't use special characters in abbreviated dialing lists used with chime paging.

- Conference — Attendant

  A call cannot be conference while the attendant is accessing paging equipment. The attendant can, however, release the call after paging the called party.

- Conference — Terminal

  A call cannot be conferenced while the user is accessing paging equipment.

- PagePac Paging Systems

  If you use chime code paging with a PagePac system, you can only page one zone at a time. PagePac systems expect a 2-digit code to access a particular zone. The system, however, immediately plays the chime code once a connection is established.

- Transfer

  A call cannot be transferred while the attendant is accessing paging equipment.

Related topics

Refer to ''Setting up voice paging over loudspeakers'' on page 463 to administer voice paging.

Refer to ''Setting up chime paging over loudspeakers'' on page 466 to administer chime paging.

Malicious Call Trace

Malicious Call Trace (MCT) allows you to trace malicious calls. MCT allows you to define a group of telephone users who can notify others in the group when they receive a malicious call. These users then can retrieve information related to the call. Using this information, you can identify the malicious call source or provide information to personnel at an adjacent switch to complete the trace. MCT also allows you to record the malicious call.

You allow users in the group to activate MCT and/or to control malicious call trace. The controlling telephone user, or controller, receives the information that MCT collects on the call.

MCT Voice Recorder

The MCT Voice Recorder is any type of audio recorder (for example, a standard audio cassette player) that you can control via the Avaya Auxiliary Trunk board.

To record the call, manually place the MCT Voice Recorder in Record mode. The telephone user then activates the MCT feature which applies power to the recorder (via the connected Auxiliary Trunk's control signal interface).

Activating MCT

To activate MCT while on an active malicious call, perform one of the following:

- Push an MCT-Activate feature button
- Place the call on hold, get a second call appearance, and dial an MCT-Activate Feature Access Code (FAC). After the dial tone, the user then dials their own extension, presses #, or waits for a 10-second time-out.
- Signal another user in the defined group to activate MCT. The co-worker activates MCT, waits for the dial tone, and dials the call recipient's extension.
- Inform a controller, who can request that another switch continue tracing the call.

  The switches must be tandemed. The controller on the first switch supplies the trunk member port id to be traced. The controller on the second switch activates MCT and presses *, followed by the trunk port id. The letters A through E of a port id are entered as 1 through 5 on the station keypad. For example, trunk port id 01C0401 would be entered as 0130401.

Once MCT is activated, information on the call is collected and alerts users in the group. The alert is not a call, so it is not affected by queues at the user's terminal. If an MCT Voice Recorder is connected, it begins recording the conversation.

NOTE:
Any Bridging, Conference, or Intrusion tone connected to parties on the connection are temporarily removed while the MCT Voice Recorder connects.

Controlling MCT

The first controlling terminal to respond to an MCT alert becomes the controller for the call. Alerting on any other controlling terminals stops.

To begin trace of an incoming call using the Malicious Call Trace feature, the controller dials the 3-digit equipment location of the incoming trunk.

During alerting, the controller's display shows the message "MALICIOUS CALL TRACE REQUEST." While this message displays, no information on incoming calls displays.

When the controller pushes the MCT-Control button, information displays identifying the called party. When the controller pushes the button again the remaining MCT information displays.

MCT collects and displays three types of information, calling information as follows:

- If the call originated inside the system or on the same node within a DCS network, the calling number displays.
- If the call originates outside the system and an ISDN calling number identification is available on the incoming trunk, then the calling number displays. Otherwise, the incoming trunk-equipment location displays. In this case, the user must call the connecting switch.

For all calls, the system displays the called number, the activating number, whether the call is active, and identification of any other parties on the call.

Deactivating MCT

The controller deactivates MCT by dialing the MCT-Deactivate FAC. Deactivation frees resources involved in the trace that were blocked. When all parties hang up, the MCT Voice Recorder disconnects.

Administering MCT for ISDN notification

The following describes how to administer the ISDN MCT notification for an ISDN trunk group (public-ntwrk, tandem, tie, or access).

Display the System-Parameter Customer-Options screen and ensure that ISDN is enabled.

One of the following must be set on the DS1 screen:

- If the DS1 is connected to the public network in Australia, set `Country Protocol` field to **2**.
- For a private network of MultiVantage systems, set `Country Protocol` to **1** and `Protocol Version` to **a.** This is recommended if DCS features are used in the private network.
- For a private network of MultiVantage systems, set `Peer Protocol` to **q-sig**. (`Peer Protocol` appears on the DS1 screen when `Signaling Mode` field is **isdn-pri**, `Connect` field is **pbx**, and `Interface` field is **peer-master** or **peer-slave**.)

One of the following must be set on the ISDN-BRI Trunk Circuit Pack screen:

- If the ISDN-BRI is connected to the public network in Australia, set `Country Protocol` field to **2**.
- For a private network of MultiVantage systems, set `Interface` to **peer-master** or **peer-slave**.

Considerations

- If the originator of the call disconnects, MCT drops; if the recipient of the call disconnects, MCT does not drop.
- Except for Emergency Access to the Attendant, features that normally display information do not do so on a controlling telephone. Otherwise, these features function normally until MCT deactivates.
- Do not use FACs to activate MCT because the process takes users too long.
- Visually Impaired Attendant Service (VIAS) voices-out display information for MCT activation, but not for MCT control.
- MCT information on an active malicious call is lost during a switch failure.
- When directing a trace to an adjacent switch, consider the following:
  - The malicious caller may hear a warning tone as a result of the intrusion.
  - You may lose continuity on the trace because the person activating MCT on the second switch may not be the MCT controller.

- If a malicious call comes in on a non-ISDN trunk, the controller needs the telephone number for the connecting switch and a cross-reference of system-trunk port numbers (including DS1 channel number, if appropriate) not the trunk equipment locations at the connecting switch. Be sure that they have this information.
- The following are the system initiated operations for MCT:
  - Conversation Recording — After the user activates MCT, the system attaches a MCT Voice Recorder, if available, to record the conversation, if available.
  - Historical Recording — After the user activates MCT, the system records the MCT-information that you can subsequently retrieve via the MCT History report.

Interactions

- Bridged Call Appearance

  If a user at a primary extension receives an indication call, then a telephone with a bridged call appearance of this extension can bridge on to the call. For an MCT-Activate button push, if the currently active extension is a bridged appearance, the system records the primary extension as the MCT recipient.

  For an MCT-Activate FAC, the user dials the number of the telephone with the bridged call appearance that is actually on the call, instead of the bridged number. However, the system logs the primary extension as the recipient. Likewise, for self-originated MCT activations via FAC, the system logs the primary extension as the MCT recipient provided that the recently held appearance is a bridged call appearance. When you activate MCT for yourself, dial # or wait for interdigit time-out.
- Conference

  A user can use conferencing to place a malicious caller on hold. The user can initiate conferencing and enter the MCT-Activate FAC, then stop conferencing and return to the malicious caller’s appearance.

  MCT-Activate can be generated for a member of a conference and is not affected by the number of parties on the conference.
- Centralized Attendant Service

  MCT-Activate, MCT-Control, and MCT-Deactivate must be performed by telephones within the same PBX.

- DCS

  If a telephone in a DCS network is involved in a malicious call, the extension is recorded and displayed with the MCT information. MCT notification passes over ISDN-PRI DCS trunks but MCT-Activate, MCT-Control, and MCT-Deactivate must be performed by telephones within the same DCS node.

- Emergency Access to the Attendant

  Ordinarily, during MCT-Control no other feature can access the controlling telephone's display. However, MCT gives up control of the display until the Emergency Access call has completed.

- ISDN

  ISDN notification of an MCT activation takes place if either the originator of the call is an ISDN trunk group with Country Protocol 2 or any trunk on the call is an ISDN private network trunk with Country Protocol 1 and Protocol Version "a" or Peer Protocol q-sig. When the ISDN trunk group is Country Protocol 2, notification is sent only to the public network.

- Make-Busy/Position-Busy/Send All Calls

  The switch attempts to activate Make-Busy or Position-Busy for telephones or consoles that activate MCT-Control. If a user has a Send All Calls (SAC) button administered but not active for the primary extension on the phone, SAC activates when the user activates MCT-Control. When the user deactivates MCT, SAC stays active unit it is deactivated by the user.

- Music-On-Hold

  If an agent places a malicious call on hold that is being recorded and the call goes to music-on-hold, the music-on-hold port and the MCT Voice Recorder port can lock. In this case, the MCT Voice Recorder continues to record the music-on-hold and is unavailable for recording subsequent malicious calls. You must perform a busy-out/release on the MCT Voice Recorder port to drop the connection.

- Priority Calling

  A priority call to an MCT recipient is denied.

- QSIG Global Networking

  MCT notification passes over the following ISDN QSIG trunk groups: tandem, tie, access, and DMI-BOS. QSIG supplementary services name and number ID provide a malicious caller's name and telephone number.

- Transfer

  If a user transfers a malicious call, the MCT information displayed on the controlling telephone identifies the transferring party as the MCT recipient.

  A user transfers a malicious caller to hold. The user initiates a Transfer, receives the second dial tone, enters the MCT-Activate FAC, then halts the remainder of the Transfer operation and returns to the malicious caller's appearance.

- Trunk Access Code

  To activate MCT for a Trunk Access Code (TAC), a user must have an MCT-Control button administered. The user hears a dial tone and enters the trunk-member number for the trunk group that the TAC identified. The user then becomes the MCT controller for a call involving the identified trunk member. This TAC operation is useful when users need to trace a call that has tandemed through their switch to terminate on another switch.

- Trunk Groups

  If a Personal Central Office Line (PCOL) is involved in an MCT, then the switch may hold up the trunk until the MCT deactivates.

Misoperation Handling

Misoperation Handling defines how calls are handled when a misoperation occurs. A misoperation can occur either:

- When a user with a call on hold goes on-hook before an operation completes. In some cases, going on-hook completes the operation, as in call transfer.
- When the system enters Night Service while attendant consoles have calls on hold.

You can alter standard Misoperation Handling to ensure that callers are not left on hold indefinitely with no way to reach someone for assistance or that callers are not dropped by the system. See Misoperation Alerting and Intercept Treatment on Failed Trunk Transfers on the Feature-Related System Parameters screen for more information.

Contact an Avaya representative for instructions on administering Misoperation Handling for use in France.

Detailed description

Misoperation handling varies considerably, depending on how it is administered and what conditions are in effect when a call is placed on hold.

Standard operation

Standard misoperation handling is in effect when you set the Intercept Treatment on Failed Trunk Transfers field to **y** and set the Misoperation Alerting field to **n**.

The type of telephone that is in use when the misoperation occurs (analog single-line or digital multiappearance) affects standard misoperation handling.

The following scenario describes a typical misoperation on an analog terminal.

1. While connected to an incoming external call, an analog-terminal user flashes with the intent of transferring the call to another terminal.
2. The user hears dial tone, dials an invalid extension, and hangs up.
3. A misoperation occurs. The analog-terminal user receives priority ringback indefinitely.

The following scenario describes a typical misoperation on a digital terminal.

1. While connected to an incoming external call, a digital-terminal user with a call on hold goes on-hook during another feature operation before completing the operation.
2. The user hears dial tone, dials an invalid extension, and hangs up.
3. A misoperation occurs unless going on-hook completes the operation (for example, call transfer). The held call remains on hold indefinitely with a flashing call-appearance lamp.

Enhanced operation

Once you enable Misoperation Alerting, calls are handled depending upon the type of call placed on hold and the type of telephone (digital multiappearance, analog single-line, or attendant console) in use when the misoperation occurs. The following list describes the 3 call types that determine misoperation handling.

- Call Type 1 — An outgoing public-network call is classified as Type 1 when answer supervision is received or when the trunk group’s Answer Supervision Timer expires, even if the trunk is still ringing. An incoming call is classified as Type 1 when it is answered.
- Call Type 2 — An incoming external-public-network call is classified as Type 2 before it is answered. A misoperation cannot occur with a Type 2 call because an unanswered incoming call cannot be placed on hold without first being answered.
- Call Type 3 — All internal calls, conference calls, and tie-trunk calls are classified as Type 3.

Analog terminal misoperation

The following 2 scenarios describe typical misoperations on an analog terminal.

Scenario 1.

1. While connected to an incoming external call (Type 1), an analog-terminal user flashes to transfer the call to another terminal.
2. The user hears dial tone, dials an invalid extension, then hears intercept tone.
3. When the user hangs up, the call re-alerts the user for 15 seconds and eventually routes to the attendant.

Scenario 2.

1. While connected to an incoming external call (Type 1), an analog station user flashes to place the call on hold while calling another extension.
2. The user hears dial tone and dials the CAS Remote Hold/Answer Hold/unhold access code.
3. The user dials an extension and talks with the user at the extension and hangs up.
4. A misoperation occurs because the first call is still left on hold.
5. The terminal is alerted for 15 seconds and the call routes to an attendant.
6. If the first call is not answered before the timer expires, the call drops.

Digital terminal misoperation

The following scenario describes a typical misoperation on a digital terminal.

1. While connected to an incoming external call (Type 1), a digital-terminal user places the call on hold to transfer the call to another terminal.
2. The user hears dial tone, dials an invalid extension, and hangs up.
3. A misoperation occurs. The call on hold rings the terminal again (not priority ringing) for the number of rings administered for call coverage.
4. The call then routes to the terminal’s coverage path, which directs the call to an announcement and/or disconnects.

Attendant console misoperation

A misoperation occurs on an attendant console with calls on hold only when the system enters Night Service.

1. The system enters Night Service with calls on hold at an attendant console.
2. All calls on hold re-alert (as if the Held-Call Timed-Reminder had expired).

3. When the calls start re-alerting, a timer starts. The timer is set to the value assigned in the Alerting (sec) field on the Console-Parameters screen.
4. If the attendant does not answer the calls before the timer expires, calls route to the system Night Service destination. Calls that are not answered at the night service destination before the night-service-disconnect timer expires are dropped.

Interactions

- Attendant Lockout

  Attendant Lockout is temporarily disabled on calls that re-alert the attendant console following a misoperation. This allows an attendant to answer the calls.

- Bridged Appearances

  Misoperation Alerting calls do not re-alert on bridged call appearances.

- Voice Response Integration

  Do not use Misoperations Handling with this feature if you are using analog boards with Conversant.

Modem Pooling

(Not available with Offer B) Modem Pooling allows switched connections between digital-data endpoints (data modules) and analog-data endpoints via pods of acoustic-coupled modems. The analog-data endpoint is either a trunk or a line circuit.

Detailed description

Data transmission between a digital data endpoint and an analog endpoint requires conversion via a modem, because the Digital Communications Protocol (DCP) format used by the data module is not compatible with the modulated signals of an analog modem. A modem translates DCP format into modulated signals and vice versa.

Modem Pooling feature provides pools of integrated-conversion modems and combined-conversion modems.

Integrated-conversion modem pools have functionality integrated on the Pooled Modem circuit pack, providing two modems. Each one emulates a time-division multiplexing (TDM) cabled to a 212 modem. Integrated are modem pools not available in countries that use A-law companding.

Combined-conversion modem pools are TDMs cabled to any TDM-compatible modem. Combined-conversion modem pools can be used with all systems.

When the system needs a modem, it queries the digital-data module associated with the call to determine if the module's options are compatible with those supported by the modem pools. If the options are not compatible, the originating user receives intercept treatment. If the options are compatible, the system obtains a modem from the appropriate pool. If a modem is not available, the user receives reorder treatment.

The system can detect the needs for a modem. Data calls from an analog-data endpoint require that the user indicate the need for a modem, because the system considers such calls to be voice calls. Users indicate this need by dialing the data-origination access code before dialing the digital-data endpoint.

The system provides a Hold Time parameter to specify the maximum time any modem can be held but not used (while a data call is in queue).

The integrated-conversion modems support the following options:

- Receiver responds to remote loop
- Loss of carrier disconnect
- Send space disconnect
- Receive space disconnect
- CF-CB common
- Speed, duplex, and synch (administered)

Combined-conversion modems support the following:

- IBM bisynchronous protocols typically used in 3270 and 2780/3780 applications. Both require 2400 or 4800 bps, half-duplex, synchronous transmission.
- Interactive IBM-TSO applications using 1200 bps, half-duplex, asynchronous transmissions
- DATAPHONE II switched-network modems supporting asynchronous and synchronous communications, and autobaud at 300, 1200, or 2400 bps
- The Avaya MultiVantage system operating at up to 19.2 kbps
- Different pools with different data-transmission characteristics

Considerations

- On data calls between a data module and an analog-data endpoint, Return-to-Voice releases the modem and returns it to the pool. The phone user connects to the analog-data endpoint.
- For traffic purposes, the system accumulates data on modem-pooling calls separate from voice calls. Measurements on the pools also accumulate.
- When a phone user places a data call to a digital-data endpoint, does not transfer the call to another digital-data endpoint, and uses a modem or acoustically-coupled modem, the user dials the data-origination access code before dialing the distant endpoint.
- Modem Pooling is not restricted. Queuing for modems is not provided, although calls queued on a hunt group retain reserved modems.
- Avoid mixing modems from different vendors within a combined pool because such modems may differ in transmission characteristics.
- When you administer data-transmission characteristics (speed, duplex, and synchronization mode), they must be identical to the TDM and optional modem selections made by the customer.
- Each data call that uses Modem Pooling uses four time slots (not just two). As a result, heavy usage of Modem Pooling could affect TDM bus-blocking characteristics.
- Tandem switches do not insert a pooled modem. The originating switch inserts a pooled modem.

Interactions

- Call Detail Recording

  Data call CDR records the use of modem pools on trunk calls.
- Data Call Setup

  Data calls to or from a TDM cannot use Modem Pooling.
- Data Privacy and Data Restriction

  The insertion of a modem pool does not turn off Data Privacy or Data Restriction.
- Data-Only Off-Premises Extensions

  Calls to or from a Data-Only Off-Premises Extension cannot use Modem Pooling, when this type of digital-data endpoint uses a TDM.

- Digital-multiplexed Interface Trunks

  If you place a data call from a local analog-data endpoint to a DMI trunk, you must dial the data-origination access code to obtain a modem. Data calls on DMI trunks to local analog-data endpoints automatically obtain modems.

- DS1 Tie Trunk Service

  Connect modems used for Modem Pooling to AVD DS1 tie trunks via Data Terminal Dialing or by dialing the feature-access code for data origination.

Related topics

To set up modem pooling for single-carrier cabinets, refer to the *Made Easy Tool for DEFINITY® Media Server Configurations*.

To set up modem pooling for compact modular cabinets, refer to *Installation, Upgrades and Additions for Avaya CMC1 Media Gateways*.

To set up modem pooling for multi-carrier cabinets, refer to *Made Easy Tool for DEFINITY® Media Server Configurations*.

To set up modem pooling for DEFINITY Server SI cabinets, refer to *Made Easy Tool for DEFINITY® Media Server Configurations*.

Multiappearance Preselection and Preference

Multiappearance Preselection and Preference selects the call appearances you use to connect to when you lift the handset.

Multiappearance Preselection and Preference provides multiappearance telephone users with the following options for placing or answering calls on selected call appearances.

- Ringing Appearance Preference

  When a user lifts the handset to answer a call, the system automatically connects them to the ringing call appearance. If there is more than one call, the user automatically connects to the oldest (first-in) ringing call appearance. The in-use (red) lamp tracks the ringing appearance and the answered appearance.

- Idle Appearance Preference

  When a user lifts the handset to place a call, the system automatically connects them to an idle appearance even if an incoming call is ringing at another appearance. The in-use (red) lamp tracks an idle appearance when the user lifts the handset.

- Last Appearance Preference

  When a user lifts the handset, they connect to the call appearance or bridged appearance last used for a call, unless an audibly ringing call on a different appearance caused the line selection to move. If the line selection moves, a call may be originated, answered, or unheld, depending on the state of that appearance.

- Preselection

  Before lifting the handset to place or answer a call, the user can press a call appearance button or a feature button to select an appearance when the in-use lamp is dark. Preselection reenters a held call or activates a feature or the speakerphone if the telephone is so equipped.

  Preselection overrides both Preference options. If the user does not lift the handset within 5 seconds after using Preselection, the selected appearance returns to idle.

  You can assign a preselection feature button. For example, if a user presses an Abbreviated Dialing button, a call appearance is automatically selected. If the user lifts the handset within 5 seconds, the system automatically places the call.

  Preference dictates whether a user connects to the ringing call appearance or to an idle call appearance. If there is no call, users automatically connect to an idle call appearance when they lift the handset, regardless of which Preference option is assigned.

Considerations

- Multiappearance telephones can have from 2 to 10 call appearances. One of these call appearances is reserved for placing calls or for receiving a Priority Calling call. If a telephone has 2 call appearances and one of them is active, a nonpriority call cannot access the other call appearance, even if the call appearance is idle. The default number of call appearances is 3.
- The reserved call appearance is not a fixed-position button; it is just the last-idle call appearance. For example, if a telephone has 10 call appearances, any 9 can be in use, but the tenth (last) is reserved.

Interactions

- Automatic Incoming Call Display

  Incoming calls are not displayed if Idle Appearance Preference is activated.

- Call Coverage

  If you administer Cover All Calls as the redirection criterion for a telephone, administer Idle Appearance Preference for the telephone. The called party can then lift the handset without accidentally connecting to a call that should be screened.

- Integrated Services Digital Network — Basic Rate Interface

  When an ISDN-BRI telephone (with Select Last Used Appearance enabled) transfers a call while off-hook by using the handset, the user hears dial tone on the last-used call appearance. Users of other telephone types hear silence in this case.

Related Topics

For more information, refer to ‘‘Station’’ on page 1066. You administer this feature with the following two fields:

- Idle Appearance Preference
- Select Last Used Appearance

Multifrequency Signaling

Multifrequency (MF) signaling is a form of address signaling used between switches and the central office (CO). It is similar to dual-tone multifrequency (DTMF) signaling in that tones convey the dialed number.

With MF signaling, the signal is typically a combination of two frequencies from a group of 5 or 6 frequencies (2/5 or 2/6). The origination switch and destination switch exchange tones that have specific meanings according to the MF protocol.

Detailed description

Avaya MultiVantage supports two frequency groups:

- R2-multifrequency compelled signaling (R2-MFC) frequency
- R1 frequency (for Spain and Russia)

R2-MFC is a version of MFC recommended by the International Telecommunication Union (ITU). It provides signaling between a CO and a switch over analog or digital CO, Direct Inward Dialing (DID), or Direct Inward and Outward Dialing (DIOD) trunks. It also provides signaling between 2 MultiVantage switches.

Avaya MultiVantage provides MF signaling that complies with ITU regulations and national regulations for specific countries. It provides these types of MF signaling: Multifrequency Espana (MFE), MF Shuttle, and multifrequency compelled (R2-MFC). These protocols signal the called number, the calling party's number (automatic number identification (ANI)), and information about the type of call or type of caller (category).

Avaya MultiVantage allows prefix digits for ANI sent on outgoing calls to be defined per PBX or per the originator's class of restriction.

If a call is a tandem call and the incoming and outgoing trunk use different protocols, the switch makes no attempt to convert between the various protocol's meanings for category. Instead,

- the PBX uses the incoming trunk's Class of Restriction (COR) assigned category if the outgoing trunk is Russian or R2-MFC, and
- the PBX uses ARS call types if the outgoing trunk is MFE.

Avaya MultiVantage provides the incoming ANI to all features on the switch that need to identify the calling party.

MFE

MFE, for Country code 11 (Spain), uses R1 frequency and compelled signaling. It is available on CO and DID trunk groups. There are four kinds of MFE signaling:

- Public 2/5
- Public 2/6
- Ibercom 2/5
- Ibercom 2/6

MF Shuttle

MF shuttle signaling, for country code 15 (Russia), uses R1 frequency and noncompelled signaling. With MF shuttle signaling, it is possible to change to decadic rotary pulse in the middle of address signal exchange. MF shuttle signaling is available on CO, DID, and DIOD trunk groups.

Also, automatic number identification (ANI) transmission, for Country code 15, uses a gapless R1 MF signal and is completed within 800ms. This is available on an outgoing CO trunk group.

R2-MFC

R2-MFC permits each country to define the meanings of the R2 frequency combinations.

R2-MFC Considerations

- Both non-group II signaling and group II signaling are supported on incoming MF signaling calls. The group II signaling protocol has an extra signal that provides caller-category information. Only group II signaling is supported on outgoing MF signaling calls.
- MF signaling also can be used in tandem trunk groups. After address signals are collected from an incoming group II MF signaling call, the call can route to a group II MF signaling trunk.
- Both incoming and outgoing MF signaling calls support ANI. The terminal displays ANI information and the Call Detail Recording (CDR) record records it.
- When Avaya MultiVantage uses an open numbering plan, the end-of-dial signal must be defined in the incoming Group I signal administration. After sending all address digits, the CO sends the end-of-dial signal to Avaya MultiVantage.
- If Avaya MultiVantage makes an outgoing call to the CO that uses an open numbering plan, the CO should send the signal A.1 to Avaya MultiVantage after sending the last address digit to the CO. Then, the CO should time-out and send a pulsed signal A.3 to Avaya MultiVantage requesting the Group II signal.
- Avaya MultiVantage offers the option to record the Calling Party Category in the call detail record (CDR). For incoming external calls, this comes from the Group II signal. For internal calls and station-originated external calls, this comes from the Class of Restriction (COR) of the originating station. For tandem calls, this value comes from the Group II signal, determined by the COR of the originating trunk group. The CDR device must be capable of receiving this information. See ‘‘CDR System Parameters’’ on page 624 for more information.
- You can assign Calling Party Category and Called Party Category on a trunk-by-trunk basis. See ‘‘Multifrequency-Signaling-Related System Parameters’’ on page 992 for more information.
- You can record and announcement to play when outgoing R2-MFC trunk calls do not complete. This applies when Avaya MultiVantage receives either group A or B signals from the called Central Office or other switch. See ‘‘Multifrequency-Signaling-Related System Parameters’’ on page 992 for more information.

Guidelines for administering MF signaling

To administer MF signaling, first you identify the origination switch and the destination switch. (The switch making the call is the origination switch; the switch answering the call is the destination switch.)

- The origination switch creates *forward* signals, classified as group I and group II signals.
- The destination switch creates *backward* signals, classified as group A and group B signals.

Group I and group A signals comprise the basic signaling for the dialed number. More elaborate signaling requires Group II and group B signals. Signal meanings and timer values can be administered.

The sequence below shows a typical interaction between the origination (forward group I and group II signals) and destination switch (backward group A and group B signals).

Forward			**Backward**
Group I	digit	-->	
	<--	A.1	Group A
	digit	-->	
	<--	A.1	
	digit	-->	
	<--	A.1	
	digit	-->	
	<--	A.1	
	digit	-->	
	<--	A.3	End of dial
Group II	II.2	-->	
	<--	B.x	Group B

Second, you assign the correlation between signal codes and their meanings.

1. Assign a code to every message. The code consists of a group category, like group II or A, and a number.
   - For example, you might assign code A.1 to the message "next-digit."
2. Assign a signal to each identifying code.
   - In every country, the frequencies (levels may differ by country) assigned to the identifying codes are the same. However, the messages assigned to the identifying codes may be different.

     For example, in Switzerland the B.6 code and its associated signal convey the *free* message, while in Thailand, *free* is conveyed by the B.1 code and its associated signal. But in both Switzerland and Thailand, the frequency associated with the B.1 code is the same.

     As another example, you might assign the signal "busy" to the B.1 code.

To receive Russian incoming ANI:

- On the DID or DIOD trunk group screen, set the Country field to **15** and the Protocol Type field to **inloc**.
- On AAR and ARS Digit Analysis Table screen, set the ANI Req field to **y** or on AAR and ARS Digit Conversion Table screen, set the ANI Req field to **y**.

Interactions

- ASAI

  ANI collected from incoming R2-MFC signaling can be used with ASAI.
- Abbreviated Dialing

  Although calls dialed automatically from an abbreviated dialing privileged list complete without class of restriction (COR) checking, ANI prefix and ANI truncation still apply.
- Attendant Console

  If the attendant assists or extends a call for a station via Straightforward Outward Completion and Through Dialing, and if the attendant has not yet released the call when the request for ANI comes in from the far end, the attendant's COR is used to select the ANI for the call. If the attendant has already released the call when the request for ANI comes in from the far end, the attendant's COR is used to select the ANI for the call.

- Authorization Codes

  The COR of the authorization code as administered on the authorization-code screen is not used for ANI prefix determination, even if the originating endpoint enters an authorization code before call processing for an outgoing call seizes an outgoing trunk. If the originating endpoint is an extension, the extension's ANI is used. If the originating endpoint is an incoming trunk, the ANI for PBX is used.

- Bridging

  The ANI of a telephone's primary extension also applies to calls originated from a bridged call appearance of that extension on another terminal. ANI prefix and ANI truncation will still apply to the primary extension number of bridged call appearances.

- Call Detail Recording

  CDR records ANI collected from incoming MF signaling.

  For India MFC, on incoming calls, ANI digits may be appended with zeroes if the actual ANI digits are less than the administered ANI-length; in those cases, the zero-digits appear on CDR.

- Call Redirection

  A call is redirected if any of the following are active: Call Forwarding, Call Coverage, Send All Calls, or Night Service.

- Call Vectoring

  Call Vectoring can now use ANI collected from incoming MFC signaling.

  The ANI of a call vector is not used when a call vectoring route-to command routes a call over an outgoing trunk. Instead, the ANI of the originating party is sent.

- DID No Answer Timer

  DID No Answer Timer is applied to MF signaling DID calls.

- Distributed Communications System (DCS)

  In a DCS arrangement, the ANI sent to the CO is determined by the ANI for PBX on PBX_B, but the category sent to the CO is determined by the Category for MF ANI field on the Class of Restriction screen for the incoming DCS trunk or by the type of call.

- Expert Agent Select (EAS)

  For ANI, the EAS agent's login extension number and COR overrides the extension number and COR of the physical terminal where the agent is logged in. ANI prefix and ANI truncation apply to logged in EAS agents.

- Hunt Groups and Automatic Call Distribution (ACD) Splits

  For ANI, a physical terminal's extension number and COR overrides the extension number and COR of the hunt group or ACD split that the terminal is a member of or logged into. ANI prefix and ANI truncation apply to terminals that are members of hunt groups or logged into ACD splits.

- Intercept treatment

  For DID MF signaling calls that are denied, you can administer whether the corresponding B.x signal or intercept tone should be sent to the CO. The default is to send the administered DID/TIE/ISDN Intercept Treatment. If the option to send the B.x signal is set, then:

  — For Group II calls, the B.x signal for the intercept is sent to the CO.

  — For non-Group II calls, if the CO dials an invalid number, the trunk is locked (regardless of this option). If the CO dials a number that is valid but not assigned, intercept tone is sent to the CO.

- Multimedia Call Handling (MMCH)

  For call origination, multimedia complexes use the COR assigned to their telephones. ANI prefix and ANI truncation will apply to the telephones assigned to multimedia complexes.

- Off-Net Call Coverage or Call Forwarding

  If the originating endpoint is an extension, the extension's ANI is used. If the originating endpoint is an incoming trunk that can supply ANI, the ANI received from the incoming trunk is used. If the originating endpoint is neither of the above, the ANI for PBX is used.

- Personal Station Access (PSA)

  For ANI, the PSA extension number and COR overrides the extension number and COR of the physical terminal where the PSA extension number is associated. ANI prefix and ANI truncation will apply to associated PSA extension numbers.

- Remote Access

  The COR of a remote access barrier code is not used for ANI prefix determination when the originating end point dials a remote access extension and then places a call. If the originating endpoint is an extension, the extension's ANI is used. If the originating endpoint is an incoming trunk, the ANI for PBX is used.

- Station Set Displays

  When no ANI is possible, if station sets are equipped with display option, they do not display the ANI digits. Instead, the trunk group name displays. When ANI is possible, ANI displays on the station set.

  For India Only. If ANI digits are padded with “zero,” then zeroes also are displayed along with ANI digits.

- Tandem / Offnet Calls

  If ANI digits are received on incoming MFC calls, the ANI digits are sent to outgoing tandem/off-net calls.

  For Russia Only. The ANI is requested on incoming trunks only when all the address digits have been collected. When the incoming trunk on a tandem call is a Russian incoming local trunk administered to collect ANI, the PBX collects all ANI digits before seizing the outgoing tandem trunk. This happens even if ARS is administered with a “min” value low enough that it would be possible to determine an outgoing route through digit analysis.

  For India Only. On an outgoing tandem-call, the default operation is to send the ANI-Not-Available forward signal if ANI is not available from the incoming trunk. However, in order to support this operation, leave the ANI for PBX field blank, and define the ANI-Not-Available signal.

Related topics

For administration settings, see ‘‘Multifrequency-Signaling-Related System Parameters’’ on page 992.

To establish MF Signaling for a trunk group, see ‘‘Trunk Group’’ on page 1167, (CO, DID, DIOC, TIE) - Incoming and Outgoing Dial Type.

To set up MFC ANI, refer to ‘‘AAR and ARS Digit Analysis Table’’ on page 559, ‘‘AAR and ARS Digit Conversion Table’’ on page 564, and ‘‘Class of Restriction’’ on page 636.

Night Service

Avaya MultiVantage provides the following Night Service features:

- Hunt Group Night Service
- Night Console Service
- Night Station Service
- Trunk Answer from Any Station
- Trunk Group Night Service

Hunt Group Night Service

Hunt Group Night Service allows an attendant or a split supervisor to assign a hunt group or split to Night Service mode. All calls for the hunt group then are redirected to the hunt group's designated Night Service extension (NSE). When a user activates Hunt Group Night Service, the associated button lamp lights.

Night Console Service

Night Console Service directs all calls for primary and daytime attendant consoles to a night console. When a user activates Night Console Service, the Night Service button for each attendant lights and all attendant-seeking calls (and calls waiting) in the queue are directed to the night console.

To activate and deactivate this feature, the attendant typically presses the Night button on the principal attendant console or designated console.

Night Station Service

Night Station Service directs incoming calls for the attendant to designated extensions. Attendants can activate Night Station Service by pressing the Night button on the principle console if there is not an active night console. If the night station is busy, calls (including emergency attendant calls) receive busy tone. They do not queue for the attendant.

When Night Station Service is active, incoming calls to the attendant route as follows:

- DID Listed Directory Number (LDN) calls route to a designated DID-LDN night extension.
- Internal calls route to the DID-LDN night extension (unless you administer the system so only DID-LDN calls can route to the LDN night extension).
- Non-DID calls route to the night destination that you specify for the trunk group or for the individual trunk. If you do not specify a night destination, the calls route to the DID-LDN night extension.

You can assign a unique extension as the night destination for each incoming central-office, foreign-exchange, or 800-Service trunk group. Both the extension assigned as a trunk group's night destination and the DID-LDN night extension can be phones or answering groups (such as DDC group, UCD group, or TEG).

Trunk Answer from Any Station

Trunk Answer from Any Station (TAAS) allows phone users to answer all incoming calls to the attendant when the attendant is not on duty and when other phones have not been designated to answer the calls. The incoming call activates a gong, bell, or chime and a phone user dials an access code to answer the call.

Users can activate TAAS if each of the following conditions is met:

- The attendant has pressed the Night button on the primary console or a user (if the switch has no attendant) pressed the Night Service button on the designated Night Service phone.
- A night console is not assigned or is not operational.
- Night Station Service is not active.

Trunk Group Night Service

Trunk Group Night Service allows an attendant or a designated Night Service phone user to assign one or all trunk groups to Night Service mode. When a user activates Night Service, trunk groups that are assigned a Trunk Group Night Service termination change to Individual Trunk Night Service mode so that calls coming into the trunk group are redirected to the group's designated Night Service extension (NSE). Incoming calls on trunk groups that are not assigned to Trunk Group Night Service are queued in the attendant queue. If the call remains unanswered during the Night Service Disconnect Timer interval, the incoming trunk disconnects.

In addition, a user can assign all the trunk groups to the night service mode at the same time. Then all the trunk groups are in the System Night Service mode. Any incoming calls made on the trunk groups are redirected to their designated NSE. To assign all the trunk groups to System Night Service, the user presses the System Night Service button on the principal attendant console or the Night Service button on a designated phone. You can assign a Night Service button to only one phone.

You can activate Night Service for specific trunk groups (Trunk Group Night Service) by pressing the individual Trunk Night Service buttons on the attendant console or on a phone. You can assign Trunk Night Service buttons on more than one phone.

Considerations

Considerations for Hunt Group Night Service

- Both Hunt Group Night Service and Trunk Group Night Service can be active at the same time. An incoming trunk call is redirected to the trunk group’s designated NSE. If this NSE is a hunt group or split that is in Hunt Group Night Service mode, the call is redirected to the Hunt Group NSE.
- Calls in progress (such as talking, on hold, or waiting in queue) on the hunt group or split are not affected when the hunt group or split is put in Hunt Group Night Service mode.
- When a hunt-group queue becomes empty, all idle members are placed in a busy condition.
- If Night Service is activated for a hunt group or split and a power failure occurs, the hunt group or split automatically returns to the Night Service mode.

Considerations for Night Console Service

- The night console must be identical to and have the same features as the primary console. A daytime console can double as the night console.
- Night Console Service calls to the attendant group are still handled by an attendant, even though the primary and daytime attendant consoles are out of service.
- Only one night console is allowed in the system. The night console can be activated only when the primary and daytime consoles have been deactivated.
- If Night Console Service is active and a power failure occurs, the system automatically returns to Night Console Service mode when it is powered up.

Considerations for Night Station Service

- When Night Station Service is active but you have not established Night Station extensions, a user can activate Trunk Answer from Any Station (TAAS).
- You can assign a Night-Serv button to either an attendant extension or a phone extension. An individual trunk group or hunt group can be put into night service by either an attendant extension or a phone extension with the necessary button. When a user presses this button to activate Night Station Service, all calls to that particular trunk group or hunt group are routed to the Night Service extension assigned to that group.

- If a trunk without disconnect supervision goes to Night Service, the system drops the trunk after a period of time to avoid locking up the trunk. The call is not routed to the DID-LDN night extension.

Considerations for TAAS

- If Night Service is active and a power failure occurs, the system, when brought back up, automatically returns to Night Service mode.

Considerations for Trunk Group Night Service

- All incoming calls on Night Service trunk groups go to the trunk group's NSE unless the trunk group member has its own Trunk Group Member Night Destination, in which case the calls are redirected to that destination instead of the trunk group's NSE.
- Calls already in progress on a trunk group (such as talking, on hold, or waiting in queue), are not affected when the individual Trunk Group Night Service or System Night Service is activated.
- Trunk Group Night Service and System Night Service work independently of one another.
  - When a user activates System Night Service, any trunks that are controlled by individual Trunk Group Night Service buttons remain in day service. Trunk groups that are not currently assigned to Trunk Group Night Service are assigned to System Night Service.
  - Trunks with individual Trunk Group Night Service can be removed from Night Service even though the rest of the system remains in Night Service.
  - When a user deactivates System Night Service, any trunks that have individual Trunk Group Night Service still active remain in night service.
  - Trunks with individual Trunk Group Night Service can be placed into Night Service even though the rest of the system remains in day service.
- If a trunk is added to a trunk group while that trunk group is in Trunk Group Night Service, the trunk is brought up in night service.
- Individual Trunk Group Night Service does not apply to DID trunk groups.
- If Night Service is activated for a trunk group, and a power failure occurs, the trunk group automatically returns to the Night Service mode.

- If for some reason, a phone with a trunk-ns button remains out-of-service after a system reboot and later comes back in service, the trunk-ns lamp shows the trunk status within 10 seconds of coming back in service. For example, a phone with a trunk-ns button may be unplugged when the system is rebooted. If the phone is plugged back in later, the trunk status is shown on the trunk-ns button within 10 seconds.

Interactions

Interactions for Hunt Group Night Service

- ACD

  When Hunt Group Night Service is active for a split and the night-service destination is a hunt group, the caller hears the first forced announcement for the original split. The system then redirects the call to the Night Service destination hunt group. When an agent in the Night Service hunt group becomes available, the call goes to that agent. If all agents in the hunt group are busy, the caller hears the following: forced or delayed first announcement, ringback, music-on-hold or silence, and a second announcement.

- Call Coverage

  Coverage takes precedence over Night Service. When Hunt Group Night Service is active, the NSE's normal coverage criteria and path apply. If the coverage path destination is AUDIX, AUDIX answers with the mail of the original hunt group. If the NSE is a hunt group or split of any type, the hunt group or split's call coverage criteria and coverage path apply. The coverage criteria and path can be different from that assigned to the phones that are members of that hunt group or split.

  If a coverage point is a hunt group or split in Night Service, the system considers the point to be unavailable and does not forward the call to the coverage point's NSE.

- Call Forwarding All Calls

  If a hunt group or split is in Hunt Group Night Service mode and the hunt group or split's NSE has Call Forwarding — All Calls active, the system forwards night-service calls terminating to that NSE to its designated call-forward extension.

  If the forwarded-to destination is a hunt group or split in Night Service mode, the system terminates the call at the forwarding extension.

Interactions for Night Console Service

- Trunk Group Night Service

  Activation of Night Console Service for the attendant consoles also puts trunk groups into night service, except those trunk groups for which you administered a Trunk Group Night Service button.

Interactions for Night Station Service

- Call Coverage

  Calls routed to the night extension via Night Station Service follow the coverage path of the night extension under all coverage criteria except Send All Calls.

  If a night extension has a coverage path in which Cover All Calls is administered, all attendant-seeking calls redirect to coverage. Changes to the protocol for handling DID-LDN calls (that is, forwarding attendant-seeking calls on or off premise from the night extension) do not work.

- Call Forwarding All Calls

  Calls redirected to the attendant via Call Forwarding All Calls do not route to the DID-LDN extension.

- Inward Restriction

  Inward-restricted phones can be administered for Night Station Service. Night Service features override Inward Restriction.

- Night Console Service

  Do not provide Night Console Service with this Night Station Service.

- Remote Access

  A Remote Access extension can be specified as the Night Station extension on an incoming, non-DID, trunk group.

- Tenant Partitioning

  Each tenant may have a designated night-service station. The system directs calls to an attendant group in night service to the night-service station of the appropriate tenant (when a night attendant is not available). When someone places an attendant group into night service, all trunk groups and hunt groups that belong to tenants served by that attendant group go into night service. In this case, the system routes incoming calls to the night-service destination of the appropriate tenant.

  Each tenant can have its own listed directory number (LDN) night destination, trunk answer on any station (TAAS) port, or night attendant.

- Timed Reminder

  Timed Reminder calls returning to a console that has been placed in Night Service and has an assigned DID-LDN night extension are not redirected to the DID-LDN night extension. Rather, they are dropped.

- Trunk Answer from Any Station

  TAAS and Night Station Service can both be assigned within the same system, but cannot be assigned to the same trunk group.

Interactions for TAAS

- Call Coverage

  If Night Station Service is active, calls that are redirected to the attendant via Call Coverage can be answered via TAAS.

- Call Forwarding All Calls

  If Night Station Service is active, calls that are redirected to the attendant via Call Forwarding All Calls can be answered via TAAS.

- Inward Restriction

  Inward-restricted phones can activate TAAS for incoming trunk calls. Night Service features override Inward Restriction.

- Night Console Service

  Do not provide a Night Console Service with TAAS.

- Night Station Service

  TAAS and Night Station Service can both be assigned within the same system, but cannot be assigned to the same trunk group. Activating Night Station Service also activates Night Service — Trunk Group for any trunk group without an individual trunk-group Night Service button.

- Tenant Partitioning

  Each tenant can have its own listed directory number (LDN) night destination, trunk answer on any station (TAAS) port, or night attendant.

Interactions for Trunk Group Night Service

- Call Forwarding All Calls

  If the individual Trunk Group Night Service mode and the trunk group’s NSE have Call Forwarding All Calls activated, the night service calls terminating to that NSE are forwarded to the designated extension.

- Forced First Announcements

  An interaction occurs with System Night Service and Forced First Announcement. For example, if hunt group A has a forced first announcement, assign the incoming CO trunk to terminate at hunt group A. Assign the incoming trunk's night-service destination to be another hunt group (hunt group B). Assign a Night Service button to the attendant.

  With night service active on the attendant, the incoming CO call routes to the night-service destination hunt group B and does not play the Forced First Announcement of the incoming destination's hunt group A.

- Listed Directory Number

  In System Night Service mode, all incoming LDN calls (except those using DID trunks) which have activated night service are redirected to their corresponding trunk group's NSE. Incoming LDN calls using DID trunks are directed to the Night Console Service, Night Station Service, or Trunk Answer From Any Station, respectively, whichever applies first. Non-LDN DID trunk calls terminate at the dialed extension.

Related topics

For Hunt Group Night Service

Refer to ''Attendant Console'' on page 595 for administering feature button assignments on an attendant console.

Refer to ''Hunt Group'' on page 840 for information on assigning the destination for calls when in a night service mode.

Refer to ''Station'' on page 1066 for administering feature button assignments on stations.

Refer to ''Setting up night service for hunt groups'' on page 214 for instructions.

For Night Console Service

Refer to ''Attendant Console'' on page 595 for information for administering feature button assignments on the attendant console.

Refer to ''Setting up night console service'' on page 206 for instructions.

For Night Station Service

Refer to ''Trunk Group'' on page 1167 for information on assigning an extension number to night service.

Refer to ''Listed Directory Numbers'' on page 964 for information on assigning an extension number to night service.

Refer to ''Attendant Console'' on page 595 for information on administering feature button assignments on an attendant console.

Refer to ''Hunt Group'' on page 840 for information on night service destination.

Refer to ''Station'' on page 1066 for administering feature button assignments on stations.

Refer to ''Setting up night station service'' on page 207 for instructions.

For Trunk Group Night Service

Refer to ''Listed Directory Numbers'' on page 964 for night service destination.

Refer to ''Attendant Console'' on page 595 for administering feature button assignments on an attendant console.

Refer to ''Station'' on page 1066 for administering feature button assignments on stations.

Refer to ''Setting up trunk group night service'' on page 213 for instructions.

For Trunk Answer from Any Station

Refer to ''Trunk Group'' on page 1167 for assignment of a blank extension number to night service.

Refer to ''Feature Access Code'' on page 752 for information on assigning a TAAS feature access code.

Refer to ''Console Parameters'' on page 660 for information on assigning an external alerting device.

Refer to ''Setting up trunk answer from any station'' on page 209 for instructions.

Off-Premises Station

Off-Premises Station allows a phone located outside of the building where the switch is located to be connected to the system. If central office (CO) trunk circuits are used, the voice terminal must be analog and must be FCC-registered or, outside the US, registered by the appropriate governmental agency. Digital communications protocol (DCP) sets can be used as off-premises terminals with the addition of the DEFINITY extender. IP stations can be set up off-premises using PPP connections. DS1 trunk service provides a digital interface for off-premises stations (also known as station-side DS1).

Off-premises stations are useful when it is necessary to have a voice terminal located away from the main location. The maximum loop distance for analog off-premises stations is 20,000 feet (6093.34 meters) without repeaters. For cabling distance information for the various voice terminal types, refer to the user guide for the appropriate telephone.

Detailed description

Off-Premises Station requires cross-connecting capabilities and one port on a Analog Line or DS1 Tie Trunk circuit pack for each interface to be provided. Not all analog lines can support an off-premises station. For information about user guide for the appropriate telephone.

NOTE:
The use of a message waiting indicator lamp on an off-premises station is not supported.

Interactions

The Distinctive Ringing feature might function improperly at an off-premises station due to the distance. However, the Distinctive Ringing feature can be disabled when the Off-Premises Station field is administered. If the Distinctive Ringing feature is not used with an off-premises station, the terminal receives 1-burst ringing for all calls.

Related topics

Refer to ''Installing home equipment'' on page 362 for information about setting up your off-premises station.

Refer to ''Configuring Avaya MultiVantage for telecommuting'' on page 351 for information about setting up telecommuting.

Refer to ''Station'' on page 1066 for information about and field descriptions on the Station screen.

PC Interface

The personal computer (PC) Interface consists of the PC/ISDN Platform product family. These products are used with Avaya MultiVantage to provide users of IBM-compatible PCs fully-integrated voice and data workstation capabilities.

Detailed description

Two groups of different configurations are available for PC Interface: group 1 uses Digital Communications Protocol (DCP) and group 2 uses the ISDN-BRI (Basic Rate Interface) protocol.

The group 1 configurations consist of DCP configurations that use a PC/PBX Interface card (formerly DCP expansion card) in the PC to link to the switch. Group 1 (shown in Figure 62 on page 1671) uses the following connections:

- The PC Interface card plugs into an expansion slot on the PC. The card has 2 standard 8-pin modular jacks (line and phone).
- The digital phone plugs into the phone jack on the PC Interface card.
- The line jack on the card provides a digital port connection to Avaya MultiVantage.
- The distance between the PC Interface card and the PBX should be no more than 1524m for 24-gauge wire or 1219m for 26-gauge wire.

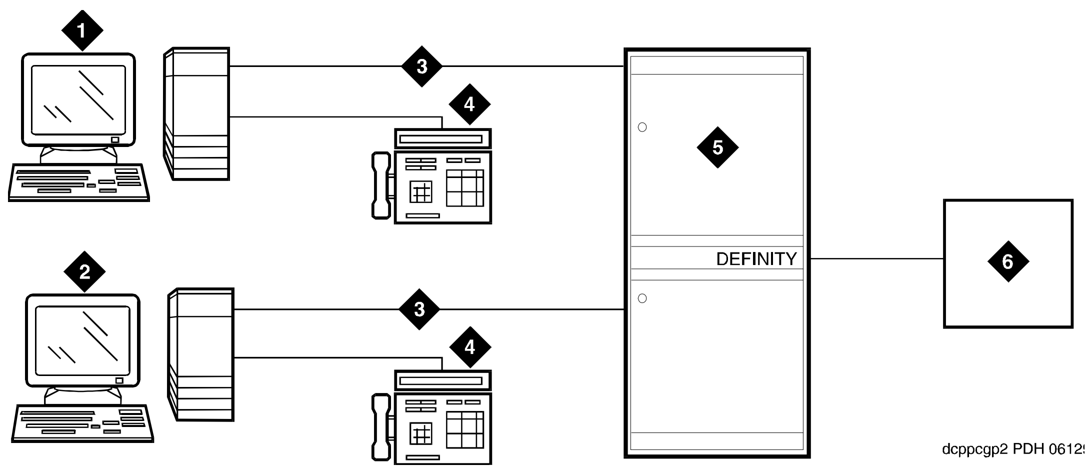


Figure Notes

1. IBM-compatible PC with DCP Interface card
2. IBM-compatible PC with DCP Interface card
3. DCP
4. DCP phone
5. Avaya Media Server (Digital Line circuit pack)
6. Host

Figure 62. DCP PC interface configuration (Group 1)

The group 2 configurations link to the switch using a PC/ISDN Interface card installed in the PC. This group can include a stand-alone PC terminal, or up to 4 phones, handsets, or headsets. Group 2 (shown in Figure 63 on page 1672) uses PC/ISDN Interface cards (up to four cards) which plug into expansion slots on the PC. These cards each provide 2 standard 8-pin modular-jack connections for both line connections (to the switch) and phone connections. A standard 4-pin modular jack is also available for use with a handset or headset.

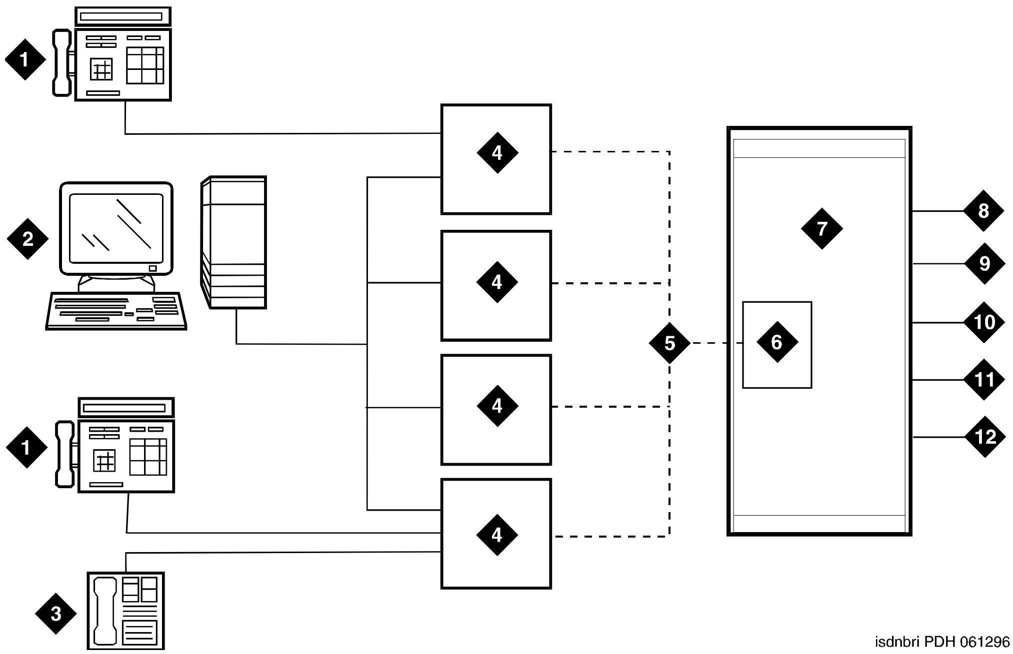


Figure Notes

1. ISDN phone
2. PC with application
3. Handset or Headset
4. BRI Interface card
5. 2B + D
6. ISDN-BRI Line, 4-wire S/T-NT, interface circuit pack
7. Avaya Media Server
8. PRI trunks
9. BRI stations
10. Interworking
11. DMI
12. Switch features

Figure 63. ISDN—BRI PC interface configuration (Group 2)

PC Interface users have multiple appearances (depending on the software application used) for their assigned extension. Designate one or more of these appearances for use with data calls. With the ISDN-BRI version, you can use up to 4 separate PC/ISDN Interface cards on the same PC. Assign each card a separate extension, and assign each extension one or more appearances. The availability of specific features depends on the COS of the extension and the COS for the switch. Modem Pooling is provided to ensure general availability of off-net data-calling services.

Security

There are two areas where unauthorized use may occur with this feature: unauthorized local use and remote access.

SECURITY ALERT:

Unauthorized local use involves unauthorized users who attempt to make calls from a PC. The PC software has a security setting so users can place the PC in Security Mode when it is unattended. You can also assign Automatic Security so that the administration program on the PC is always active and runs in Security Mode. This mode is password-protected.

SECURITY ALERT:

Remote access involves remote access to the PC over a data extension. Remote users can delete or copy PC files with this feature. You can password-protect this feature. Consult the Avaya Products Security Handbook *for additional steps to secure your system and to find out about obtaining information regularly about security developments.*

Considerations

- Use the Function Key Module of the 7405D with PC Interface.
- BRI terminals normally are initializing terminals and require you to assign an SPID. The PC/ISDN Platform (Group 2), in a stand-alone configuration, is a noninitializing BRI terminal and does not require you to assign a SPID.
  - Set a locally-defined terminal type with General Terminal Administration
  - Define the terminal type as a noninitializing terminal that does not support Management Information Messages (MIM).
  - Assign the PC/ISDN Platform with an associated (initializing) ISDN-BRI phone (such as an ISDN 7505) using a SPID.
  - Assign the station (using a locally-defined terminal type) to take full advantage of the capabilities of the PC Interface. This terminal type is also noninitializing with no support of MIMs.
- Do not use phones with data modules with the PC Interface. (You can still use 3270 Data Modules if you also use 3270 emulation). If you attach a DCP data module or ISDN data module to a phone that is connected to a PC Interface card, the data module is bypassed (not used). All the interface functions are performed by the interface card even if a data module is present.
- The 7404D phone with messaging cartridge cannot be used with PC Interface. However, the 7404D with PC cartridge can be used, but only with Group 1 configurations.

Interactions

- Data Communications Access

  PC Interface uses a digital interface and is not directly compatible with Data Communications Access, which uses an analog interface. Apply Modem Pooling conversion if you use these features together.

- Data Protection

  Assign Data Protection—Permanent for use with PC Interface for data communications.

- Host Computer Access

  Both PC Interface and Host Computer Access use digital interfaces. These features are compatible and Modem Pooling conversion is unnecessary.

- ISDN-BRI

  Each card can have its own separate phone or voice-calling device. A phone does not require special application software on the PC. However, a handset or headset alone requires special application software.

- Modem Pooling

  Use Modem Pooling if you use PC Interface to place calls to, or receive calls from, off-premises stations over analog trunks.

Personal Station Access

Personal Station Access (PSA) allows users to associate the preferences and permissions of their telephone with any other telephone of the same type.

PSA makes it convenient for different users to use the same bank of phones at different times. For example, several telecommuting or hoteling employees can use the same office on different days of the week. The employees use PSA to “associate” with the office phone — that is, make the terminal “theirs” for the day. Calls an employee originates from the station are recognized and displayed as the employee’s calls, and calls routed to the employee’s extension route to the voice terminal “associated” with that extension.

A telecommuting or hoteling employee can also use PSA when working at home. For example, the employee installs a DCP terminal and a DEFINITY Extender at home, calls into the system, and uses PSA to associate the remote phone with their extension. The system associates the home terminal — that is, recognizes the home terminal as having the employee’s preferences and permissions. When someone calls the employee’s extension, the call rings at the employee’s home. When the employee no longer needs to use the office, they “dissociate” from the terminal.

PSA requires users to enter a security code and can be used on-site or off-site. PSA-invalid attempts generate referral calls and are recorded by Security Violation Notification (SVN) software, if that feature is enabled. If a user interrupts the PSA dialing sequence by pressing the release button or by hanging up, the system does not log the action as an invalid attempt.

Detailed description

The preferences and permissions that are retained with PSA include the definition of terminal buttons, abbreviated dial lists, and class of service (COS) and class of restriction (COR) permissions assigned to the your station. Extensions that do not have a COS, such as Expert Agent Selection (EAS) agents or hunt groups, cannot use PSA.

PSA functions only on analog, hybrid, and Digital Communications Protocol (DCP) terminals. Many types of DCP terminals exist, and these terminals have different types and numbers of buttons. If you attempt to associate DCP stations with DCP terminals that have incompatible buttons, button mapping is unpredictable. This is also true of hybrid terminals. If you want a user to be able to use the terminal buttons and to have consistent displays, associate stations with terminals of the same type.

Stations and ports on different switches cannot be PSA associated. This includes stations on different switches (or nodes) within Distributed Communications The system does not limit the number of stations that can use PSA. However, heavy use of the associate and dissociate functions may temporarily impact system performance.

When a call that goes to coverage from a PSA-disassociated extension, the switch sends a message to the coverage point indicating that the call was not answered. If the coverage point is a display phone, the display shows “da” for “don't answer.” If the coverage point is a voice messaging system, the VM system receives an indication from the switch that this call was not answered, and treats the call accordingly.

Dissociated telephones

When a user requests to associate a telephone with PSA, any other telephone using that extension is automatically dissociated. It is possible to place emergency calls from a dissociated telephone, provided a COR has been assigned to dissociated phones on the Feature-Related System Parameters screen.

PSA allows a dissociate request from a bridged appearance. However, when you execute a dissociate command from Terminal B, even if you are on a bridged appearance of an extension belonging to Terminal A, you dissociate the station belonging to Terminal B.

The dissociate function within PSA allows a user to restrict the features available to a phone. When a phone has been dissociated using PSA, it can be used only to call an attendant, to place an emergency call, or to accept a Terminal Translation Initialization (TTI) or PSA request. To enable users to make other types of calls from dissociated sets, you must establish a class of restriction for these phones. See ‘‘Setting up Personal Station Access’’ on page 354 for more information.

NOTE:
Once a station has been associated with a terminal, anyone using the terminal has the capabilities of the associated station. Be sure to execute a dissociate request if the terminal can be accessed by unauthorized users. This is particularly important if you use PSA and DCP extenders to permit remote DCP access.

Interactions

- Adjunct/Switch Application Interface

  An ASAI link cannot use this feature because ASAI uses a BRI port. Do not assign to an ASAI link a Class of Service that allows PSA.

- Bridged Appearance

  When you execute a PSA dissociate request for the principal station, its bridged appearances remain active if the stations on which they appear have not been dissociated. When a call is made to the principal extension, any of its bridged appearances that can be alerted are alerted. Otherwise, the call follows the principal extension’s coverage path.

  PSA dissociate requests executed at a bridged appearance dissociates the station that the bridged appearance is on.

- Call Management

  PSA dissociate automatically logs out an ACD agent.

- Coverage

  PSA does not change coverage path operations. If a station is dissociated, its calls still go to coverage unless they are forwarded.

- Property Management System

  A station assigned to a room, rather than to a person who needs to work in multiple locations, should not use PSA. Such a station should not have PSA in its COS.

- Save Translations

  PSA commands cannot be successfully executed during a save translations. When a reset 3 or greater (reset 4, reset 5, and so on) occurs on the system, all associations revert to their state as of the last save translations.

- Security Violation Notification

  PSA security violations are tracked and reported by SVN, if it is active.

- Tenant Partitioning

  If a terminal is already associated, a user attempting a PSA associate request at that terminal must specify a station in the same partition as the station already associated with the terminal.

  However, anyone, in any partition, can execute a PSA dissociate request at the terminal (if the associated station has PSA in its COS) and then execute a PSA associate request for a station in any tenant partition.

Related topics

Refer to ‘‘Class of Service’’ on page 650 for information about and field descriptions on the Class of Service screen.

Refer to ‘‘Configuring Avaya MultiVantage for telecommuting’’ on page 351 for information about setting up telecommuting.

Refer to ‘‘Feature Access Code’’ on page 752 for information about and field descriptions on the Feature Access Code screen.

Refer to ‘‘Feature-Related System Parameters’’ on page 766 for information about and field descriptions on the Feature-Related System Parameters screen.

Refer to ‘‘Setting up Personal Station Access’’ on page 354 for information about associating the preferences and permissions on your station with any other compatible terminal.

Refer to ‘‘Setting up remote access’’ on page 364 for information about access the system from a remote location.

Refer to ‘‘Station’’ on page 1066 for information about and field descriptions on the Station screen.

Priority Calling

Priority Calling provides a special type of call alerting between internal telephone users, including the attendant. The called party receives a distinctive ring when the calling party uses Priority Calling.

You administer the priority-calling ringing-pattern system wide. Default is a 3-burst alerting signal. You allow feature use for each telephone user by administering the user’s class of service.

The following types of calls are always priority-calling calls:

- Call coverage consult
- Automatic callback
- Ringback queuing
- Attendant intrusion
- Security violation notification

The system generates the call waiting ringback tone that a single-line telephone user hears even if the user is active on a call. In contrast, the system *does not* generate the pattern for a multiappearance telephone if there are no idle call appearances. In this case the caller hears busy tone. However, the system does generate the pattern if the telephone has an idle call appearance, including the one reserved for call origination.

Interactions

- Abbreviated Dialing

  If a user wants to make a priority call to a number in an abbreviated dial list, the Priority Calling access code and the AD code must be programmed on one button. If a user attempts to use Abbreviated Dialing (either by button or FAC) after dialing the Priority Access Code, the system denies the call.

- Call Coverage

  Priority Calling calls do not redirect to coverage unless the caller activates Go to Cover. If the call redirects, it remains a priority call, and the covering user receives a distinctive (default is three-burst) ringing signal.

- Call Forwarding All Calls

  Priority Calling calls (except callback calls) are forwarded, and the forwarded call remains a priority call.

- Call Vectoring

  The system generates intercept tone when someone attempts to activate Priority Calling toward a Vector Directory Number (VDN).

- Call Waiting

  A Priority Calling call waits on an active single-line telephone even if Call Waiting is not assigned to the telephone. The active, single-line telephone user receiving the call hears a distinctive (default is three-burst) priority Call Waiting tone.

- Consult

  A Consult call acts as a priority call and waits at a single-line telephone, even if the telephone does not have Call Waiting Indication assigned.

- Distributed Communications System

  On a DCS tandem call to a single-line telephone, the called party does not receive priority ringing if the caller activates Priority Calling by pressing the priority button *after making a call*.

- Last Number Dialed

  If a priority call is to be made to the last number dialed, the Last Number Dialed button must be used. The Last Number Dialed feature access code is not valid after Priority Calling has been activated.

  Single-line telephones (2500 series) can be administered so that distinctive signals are not provided. In this case, 1-burst ringing is provided for priority calls.

Related topics

To administer Distinctive Audible Alerting settings, see ''Feature-Related System Parameters'' on page 766.

To allow Priority Calling, see ''Class of Service'' on page 650.

Recorded Telephone Dictation Access

Recorded Telephone Dictation Access permits phone users, including Remote Access and incoming tie-trunk users, to access dictation equipment.

Users start by dialing an access code or extension. Start/stop is controlled by voice or dialing. Initial activation and playback are controlled by dial codes.

Recorded Telephone Dictation Access cannot be used with the following features:

- Automatic Route Selection
- Conference

Related topics

Refer to "Audible Message Waiting" on page 1362 for information about letting users know they have messages.

Refer to "Announcements" on page 1348 for information about the messaging server interface.

Refer to "Voice Message Retrieval" on page 1798 for information about retrieving messages.

Refer to "Voice Messaging Systems" on page 1800 for more information about voice messaging systems.

Refer to "Trunk Group" on page 1167 for information about and field descriptions on the CPE Trunk Group screen. Complete all fields on this screen to administer the recorded telephone dictation access.

Refer to "Station" on page 1066 for information about and field descriptions on the 2500 Analog phones screen. Complete all fields on this screen to administer the recorded telephone dictation access.

Remote Access

Remote Access permits authorized callers to access the system via the public network from remote locations and then use its features and services. The Remote Access caller must use a touch-tone phone or equivalent equipment. Since the system does not have access to the calling (outside) number, Ringback Queuing and Automatic Callback cannot be used on a Remote Access call. Also, any feature requiring recall dial tone (for example, Hold and Transfer) cannot be accessed remotely.

Read the information in "Security" on page 1682 before administering this feature.

SECURITY ALERT:

Avaya has designed the Remote Access feature incorporated in this product that, when properly administered by the customer, enables the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes, and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed on the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of remote-access features. In such an event, applicable tariffs require that the customer pay all network charges for traffic. Avaya cannot be responsible for such charges, and does not make any allowance or give any credit for charges that result from unauthorized access.

Detailed description

Remote Access provides you with access to the system and its features from the public network. This allows you to make business calls from home or use Recorded Telephone Dictation Access to dictate a letter. If authorized, you can also access system features from any on-site extension.

With Remote Access, you can dial into the system using Direct Inward Dialing (DID), Central Office (CO), Foreign Exchange (FX), or 800 Service trunks. When a call comes in on a trunk group dedicated to Remote Access, the system routes the call to the Remote Access extension you have assigned. If DID is provided and the Remote Access extension is within the range of numbers that can be accessed by DID, Remote Access is accessed through DID.

Upon access to the feature, you hear system dial tone. If you have administered your system to require a barrier code or authorization code, the system requires you to enter it now. When you enter the required codes, the system generates dial tone. You can now place local or long-distance calls as allowed.

Barrier codes provide your system security and define calling privileges through the administered class of restriction (COR). You can administer up to 10 barrier codes, each with a different COR and class of service (COS). Barrier codes can be from 4 to 7 digits, *but all codes must be the same length*. You can also require that users enter an authorization code to use this feature. Both barrier codes and authorization codes are described under ''Security'' on page 1682.

The destination of incoming non-DID trunk calls can be an attendant or an extension. The destination is specified on each individual trunk group. When the trunk group is dedicated to Remote Access, the Remote Access extension is specified. In this case, you do all dialing. If an attendant is needed on a call, you dial the public network telephone number assigned, the barrier code, and the attendant access code. You can administer your system to provide attendant-assisted calling during the day but Remote Access after normal business hours. You do this by setting the trunk group Incoming Destination field to the attendant (attd), and specifying the Remote Access extension as the Night Service extension .

After a Digital Terminal Data Module’s (DTDM) baud rate is changed from 9600 to 1200, the DTDM cannot be accessed by Remote Access until an internal call is made to the DTDM.

Security

The system provides the ability to check the status of the remote access feature and barrier codes. The **status remote-access** command displays information that can help in determining why and when use of Remote Access or a particular barrier code was denied. The display indicates if Remote Access is:

- Not administered
- Enabled
- Disabled
- Disabled following detection of a security violation

It also gives the date and time Remote Access was last modified.

For each barrier code, the command displays:

- Date the code was administered, reactivated, or modified
- Expiration date
- Number of calls that can be placed with the code
- Number of calls that have been placed using the code
- Whether the code is active or expired
- Date and reason a code expired

For a detailed description of the status remote-access command and display, refer to *Avaya Products Security Handbook*.

Barrier Codes

Remote Access has inherent risks; it can lead to large-scale unauthorized long-distance use. To increase your system’s security, use a 7-digit barrier code with Remote Access Barrier Code Aging. You can administer the Remote Access Barrier Code Aging feature to:

- Limit the length of time an access code remains valid
- Limit the number of times an access code can be used
- Both of the above

You must administer expiration dates and access limits for each of the possible 10 barrier codes. If your system has more than 10 Remote Access users, they must share codes. A barrier code automatically expires if an expiration date or number of accesses has exceeded the limits you set. If both a time interval and access limits are administered for a barrier code, the barrier code expires when one of the conditions is satisfied.

If barrier codes are administered, a special answer-back tone causes a calling modem to leave dial mode. A modem's dialer is sometimes used to gain access (this tone also cancels echo suppressors in the network, preventing DTMF tones from breaking dial tone from a switch). Barrier codes can be used alone or with authorization codes.

To view the status of a Remote Access barrier code, use the status remote-access command.

NOTE:
Barrier codes are *not* tracked by Call Detail Recording (CDR). Barrier codes are incoming access codes, whereas, authorization codes are primarily outgoing access codes.

When you no longer need a barrier code, remove it from the system. Barrier codes should be safeguarded both by you and their users.

Authorization Codes

You can also administer authorization codes to manage access to your system. You can then use CDR to track this code use. Use these guidelines to manage your system's authorization codes.

- Assigning codes — Create random codes; do not allow them to follow a predictable pattern. Use the maximum code length allowed and assign a unique code to each person responsible for protecting the code.
- Changing codes — Change codes often.
- Deleting codes — Delete codes when they are no longer needed.
- Monitoring codes — Use CDR reports to analyze code use.

Alternate Facility Restriction Levels

Consider changing facility restriction levels (FRL) with alternate facility restriction levels (AFRL) after normal business hours to restrict where calls can be made over your facilities. Take care not to restrict callers from summoning emergency services after hours.

Class of Restriction

The COR of an authorization code supersedes that of a barrier code.

- Time of Day Routing — Controlled by the time-of-day entries in COR or by the partition.
- Toll Restriction and Analysis — Controlled by COR.
- Trunk Access Code — Interacts with toll restriction. You can translate your switch so users can make toll calls via Alternate Route Selection (ARS) without using a trunk access code.
- Trunk Administration — Remote Access trunks can be restricted.

For additional steps to secure your system and to find out about obtaining security information on a regular basis, refer to the *Avaya Products Security Handbook*.

Logoff Notification

Use Logoff Notification when Remote Access is enabled, but not actively used. Logoff Notification notifies you at logoff that Remote Access is enabled. It guards against inadvertently leaving Remote Access active and can also alert you to unauthorized feature activation. Logoff Notification is administered by login ID.

Interactions

- Authorization Codes

  When a Remote Access caller dials the assigned Remote Access extension and connects to the system, the system may request the caller to dial an authorization code in addition to a barrier code. Dial Tone between the barrier code and authorization code is optional. Calling privileges associated with the COR assigned to the authorization code supersede those assigned to the barrier code.
- Class of Restriction

  COR restrictions do not block access to the Remote Access feature.
- Night Service

  The Remote Access extension can be specified as the Night Service extension on an incoming, non-DID, trunk group.

Related topics

Refer to ‘‘QSIG to DCS TSC Gateway screen’’ on page 1029 for information about and field descriptions on the Remote Access screen.

Refer to ‘‘Setting up remote access’’ on page 364 for step-by-step instructions for configuring remote access.

Refer to ‘‘Authorization Code — COR Mapping’’ on page 611 for information about and field descriptions on the Authorization Code screen.

Refer to ‘‘Trunk Group’’ on page 1167 for information about and field descriptions on the Trunk Group screen.

Reset Shift Call

Reset Shift Call allows users to redial a local or private network call by pressing a single digit. The switch plays reset shift dial tone on busy calls if:

- A station or attendant originates a call
- The dialed endpoint is in the dial plan and is one of the following types:
  - Extension (local or uniform dial plan (UDP))
  - Prefixed extension (local or UDP)
  - AAR
- The destination is on the same switch as the originator, or is connected via an ISDN trunk to the originating switch

When the caller hears reset shift dial tone, they can press a single digit that replaces the last digit of the originally-dialed destination and the call transfers to the new destination. This feature is useful, for example, where extensions are assigned sequentially to functional organizations.

The originator and destination of the call both must be connected to Avaya MultiVantage. In DCS or QSIG environments, both must be on the same DCS or QSIG networks, but do not need to be connected to the same Avaya MultiVantage.

Interactions

- Automatic Alternate Routing (AAR)

  If a user dials the AAR feature access code followed by destination digits to place a call and the destination is busy, reset shift dial tones is heard. If the user then dials a new destination last digit, a new call is placed and the display updates as if the AAR feature access code, followed by the new destination, was dialed originally.

- Bridged Call Appearance

  If set A has a bridged appearance on set B, and both users A and B are off-hook on that bridged appearance, and set A calls set C, and set C is busy, both A and B hear reset shift dial tone. If set B then presses a single digit, the digit is ignored and both A and B continue to hear reset shift dial tone. Set A must dial the new destination last digit.

- Distributed Communications System

  Reset shift call works in a DCS network over ISDN trunks. DCS+ and QSIG networks always support reset shift calls.

- Forwarding

  If the called destination is forwarded to station A and station A is busy, the switch applies reset shift dial tone.

  If a user dials to a forwarded set and then gets reset shift dial tone, and enters a digit, the number dialed by the reset shift call feature is 1 digit off from the original user-dialed number, not the forwarded-to number.

- Hot Line Service

  If a station set user with hot line service reaches a busy endpoint, the user hears busy tone and not reset shift dial tone.

- Intercom

  - Dial

    If a user lifts the handset, presses the DIAL INTERCOM button, dials the 1-digit or 2-digit intercom code assigned to another set, and reaches a busy set, the switch plays busy tone, not reset shift dial tone.

  - Automatic

    If a user lifts the handset, presses the AUTOMATIC INTERCOM button, and reaches a busy set, the switch plays busy tone, not reset shift dial tone.

- Last Number Dialed

  If a user hits a LAST NUMBER DIALED button while listening to reset shift dial tone, the button press is denied.

  If a user uses last number dialed after having used reset shift call on the last call, the switch re-attempts the last special-dialed call.

- Line Lockout

  An analog user listening to reset shift dial tone is not subject to the line intercept tone timer.

- Line Side DS1

  A call originated from a line side DS1 station (ds1fd, ds1sa, ops, vrufd, vrusa) is considered a station originated call.

- Multimedia Call Handling

  If a user at a multimedia complex presses the MM-CALL button or dials the multimedia-origination feature access code after receiving dial tone, and then reaches a busy set and uses the reset shift call feature, the resulting new call attempt will be a multimedia call attempt.

- Priority Calling

  If a user dials the priority calling feature access code or presses a priority call button, dials a busy station, and then uses the reset shift call feature, the switch makes a priority call to the substituted destination.

- QSIG

  Reset shift call does work in a QSIG network if the QSIG network configuration allows the ISDN busy signaling to always propagate back to the originating switch, which may not be the case if the network includes non-Avaya trunks and/or switches.

- Remote Access

  A call originated from a remote access extension does not receive reset shift dial tone.

Remotely Readable Electronic Phone IDs

Remotely readable electronic identification allows the switch to read a serialized DCP telephone’s part number (comcode) and serial number. Avaya can use this information to determine the proper maintenance replacement unit or the date of manufacture for warranty or QPPCN status.

The remotely readable electronic identification is currently only available with the 6400 Serialized DCP telephones. The 6400 Serialized phone is stamped with the word “Serialized” on the faceplate for easy identification.

The remotely readable electronic ID allows both you and Avaya service personnel to obtain more accurate information about the phone, thus simplifying the repair process. With this new functionality, the serial numbers for phones can be entered into trouble tickets manually so that Avaya can identify the problem more easily. In addition, this feature enables the switch to read the model number of DCP phone types whose electronic IDs are not remotely readable.

Phones with remotely readable IDs show whatever information is available for phones connected to a DEFINITY Extender, provided that the Extender has an active connection with the switch.

Ringer Cutoff

Ringer Cutoff allows multiappearance telephone users to turn audible ringing signals on and off. This feature does not affect visual alerting.

When this feature is enabled, only Priority ring (by default 3-burst), Redirect Notification, Intercom ring, and manual signaling ring at the telephone. Internal and external calls do not ring.

The following table summarizes which calls are affected by Ringer Cutoff.

	Redirect notification is	
Call Type	**Inactive**	**Active**
telephone to telephone	no	ring ping
Attendant to telephone	no	ring ping
Internal tie to telephone	no	ring ping
APLT trunk to telephone	no	ring ping
Trunk to telephone	no	ring ping
Priority call to telephone	yes	yes
Intercom call to telephone	yes	yes
Manual signaling	yes	yes

NOTE:
If Call Coverage is set to Cover All and Ringer Cutoff and Redirect Notification are both active, then Redirect Notification is received. If Redirect Notification is not active, no audible alerting is received.

A user may not wish to be disturbed by the arrival of incoming calls, yet not want calls to be redirected immediately to coverage. For example, an executive may want a secretary to answer calls before they redirect to coverage. The bridging user (the secretary) is not affected by the executive's activation of Ringer Cutoff.

If a primary extension and all other users with bridged appearances of the primary extension activate Ringer Cutoff, an incoming call silently alerts all of the telephones and then redirects to coverage.

Interactions

- Automatic Callback

  If Ringer Cutoff is active, an Automatic Callback call returns to the user's telephone with the normal 3-burst ring.

- Bridging

  A bridging user is not affected by a primary extension's activation of Ringer Cutoff. Nor is the primary extension affected by the bridging user's activation of Ringer Cutoff.

- Call Forwarding All Calls

  If Ringer Cutoff and Call Forwarding All Calls are active, the user receives redirect notification, if you set Redirection Notification to **y** for the extension.

- Distinctive Ringing

  Ringer Cutoff turns off only the distinctive ringing of internal and external calls. Intercom ringing, priority ringing, redirect notification, and manual signaling are not turned off.

- Intercom

  If Ringer Cutoff is active, intercom calls still ring the user's telephone.

- Manual Signaling

  If Ringer Cutoff is active, Manual Signaling still rings the user's telephone.

- Ringback Queuing

  If Ringer Cutoff is active, the return call for Ringback Queuing still rings the user's telephone with a 3-burst alerting signal.

- Priority Calling

  If Ringer Cutoff is active, priority calls still ring at the user's telephone.

- Send All Calls

  When Ringer Cutoff and Send All Calls are both active, the user receives redirect notification, when Redirection Notification is set to **y** for that extension.

Ringing — Abbreviated and Delayed

Ringing — Abbreviated and Delayed allows you to assign one of four ring types to each call appearance on a telephone. Whatever treatment is assigned to a call appearance is automatically assigned to each of its bridged call appearances.

Detailed description

Ring types fall into two categories:

- Those that alert consistently and don't change:
  - Ringing, in which the lamp flashes and audible ringing occurs
  - Silent ringing, in which the lamp flashes and audible ringing does not occur
- Those that transition from one ringing state to another:
  - Abbreviated ringing, in which ringing continues for as many cycles as specified by the automatic abbreviated/delayed transition interval and then changes to silent alerting
  - Delayed ringing, in which visual alerting continues for as many cycles as specified by the automatic abbreviated/delayed transition interval and then changes to ringing

For a station with call appearances that have either abbreviated or delayed ringing, an abbreviated-ring button associated with that station's extension can be assigned on a different station. When one of those call appearances is being alerted, pressing the button forces immediate transition of the alerting — that is, from ringing to silence or from silence to ringing.

This feature is most useful in bridging situations in which some users want to be:

- Audibly alerted to a call immediately upon its arrival
- Audibly notified if the call has not been answered within a specified number of rings or if they have indicated.
- Able to stop the audible alerting if the call is not being answered by the principal and the user is not able to answer the call.

Because ring type can be specified on a per-station basis, mixing ring-type specifications within a station's access to a particular extension is possible.

For Ringing — Abbreviated and Delayed, each call appearance must be:

- Assigned a ring type
- Administered to transition when the:
  - — Abbreviated/delayed transition interval is reached or when the user presses the abbreviated ring button
  - — user presses the abbreviated ring button, regardless of the abbreviated/delayed transition interval

Considerations

- You cannot assign Ringing — Abbreviated and Delayed to an attendant console.
- You can assign Abbreviated and Delayed Ringing to analog stations. However, because analog stations cannot visually alert, a user may unexpectedly answer an incoming call while intending to originate a call.

Interactions

- Call Coverage

  If the number-of-rings interval for coverage is shorter than the automatic transition interval, a call redirects to coverage before audibly alerting a call appearance that has delayed ringing. However, timing continues for the automatic transition interval in case no coverage point is available and the call continues to alert at the station.

  When a call is immediately redirected to coverage, the Abbreviated and Delayed ringing has no effect.

- Call Forwarding — Busy/Don't Answer

  When a call is forwarded because it is not answered in the specified time, the call stops alerting the station and is not affected by the Ringing — Abbreviated and Delayed feature. However, timing continues for the automatic transition interval in case forwarding fails and the call continues to alert at the station.

  If the call forward don't answer interval is shorter than the automatic transition interval, the call redirects to the forwarded-to extension before ringing a station with a ring type of delayed ringing.

- Call Vectoring — Expert Agent Selection — Logical Agents

  Calls routed to a logical agent use the translations for the Ringing — Abbreviated and Delayed feature of the station being used by the agent.

- Data Extension Calls

  Data Extension calls are not affected by the ring values, but continue to be directed by the bridged call alerting administration.

- Hospitality Features — Do Not Disturb

  The Do Not Disturb feature takes precedence over the Ringing — Abbreviated and Delayed feature in blocking ringing to the station.

- ISDN — World Class Basic Rate Interface

  Several of the protocol variations supported by the World Class BRI feature do not permit the messaging required for control of the station's ringer by Ringing — Abbreviated and Delayed. In this case, ring type is forced to a value of ring.

- Multiappearance Preselection and Preference

  The system automatically selects any alerting call on a station whether or not it is ringing if the Per Button Ring Control field is set to **n**. If the field is set to **y**, it selects only audibly ringing calls.

- Off-Premises Station and Off-Premises Extension Lines

  You must use ring type of "ring" for OPS and OPX lines.

- PCOL Calls

  PCOL calls are not affected by the ring values, but continue to be directed by the bridged call alerting administration.

- Redirection Notification

  If Redirection Notification is enabled, terminals only receive redirection notification if the alerting button or the first call appearance has an assigned ring value of ring or abbreviated ring.

- Terminating Extension Group Calls

  TEG calls are not affected by the ring values, but continue to be directed by the bridged call alerting administration.

- Voice Mail Systems

  Voice mail systems may look for ringing applied to a port to trigger call answer. Undesirable adjunct operation may result if ring-type translations are inappropriately set for ports serving these adjuncts.

Security violations notification

When a security violation occurs, security violations notification (SVN) notifies a designated referral point. This can be an attendant console, a display-equipped phone, or a phone without display for SVN referral calls with announcements.

The system monitors and reports on the following types of security violations:

- Login violations
- Remote access barrier code violations
- Authorization code violations
- Station security code violations

Avaya MultiVantage provides the option to log a major alarm if a security violation occurs involving an Avaya services login ID. Avaya is responsible for retiring the alarm.

Refer to *Reports for Avaya MultiVantage Software* for more information on how to run reports, and respond to security violations.

To effectively monitor the security of your system, you need to know how often both valid and invalid attempts at system entry are normally made. Then you will know if the number of invalid attempts is unusually high. A significant increase in such attempts can mean the system is being compromised.

NOTE:
It is recommended that you print and clear the security-violation measurement reports at least once a month. In a busy system, once a week is not too frequent.

Security violation thresholds and notification

As an example, you may determine that during a forty-hour week, it's normal for users to submit about 1,000 valid barrier codes and 150 invalid barrier codes; that is, about 3.75 invalid barrier codes are submitted per hour.

With this information, you may decide to declare that a security violation occurs during any hour in which 8 invalid barrier codes are submitted. If you know that during an 8-hour period, about 30 invalid codes are submitted, you might set the threshold to count a security violation when 40 invalid codes are submitted within eight hours.

You can administer SVN to place a referral call to the location of your choice whenever the established thresholds are reached. All SVN referral calls are priority calls.

Invalid attempts accumulate at different rates in the various security arenas (login, authorization code, remote access, and station security code), depending on feature usage and the number of users on a server. For this reason, you administer thresholds separately for each type of violation.

Sequence of events

The following is the sequence of events that occur when an SVN is enabled and a detects a security violation:

1. SVN parameters are exceeded (the number of invalid attempts permitted in a specified time interval is exceeded).
2. An SVN referral call (with announcements, if assigned) is placed to a designated point, and SVN provides an audit trail containing information about each attempt to access the switch.
3. SVN disables a login ID or Remote Access following the security violation.
4. The login ID or Remote Access remains disabled until re-enabled by an authorized login ID, with the correct permissions.

Reporting

The system reports information about security violations in the following ways:

- **In real time** — you can use the **monitor security-violations** command to monitor security violations as they may be occurring. Enter this command, followed by the type of security violation you want to monitor (logins, remote-access, authorization-codes, or station-security-codes).
- **On an immediate basis** — when a security violation occurs, the system sends a priority call to a designated referral point (attendant console or phone). Thus, there is some chance of apprehending the violator during the attempted violation.

  Upon notification, you can request the Security Violations Status Reports , which show details of the last 16 security violations of each type. The Barrier Code and Authorization Code reports also include the calling party number from which the attempt was made, where available.
- **On a historical basis** — the number of security violations of each type, as well as other security measurements, are collected and displayed in the Security Violations Summary and Detail reports. These reports show summary information since the counters were reset by the **clear measurements security-violations** command or since system initialization. They do not show all aspects of the individual security violations.

SVN- halt buttons

You can administer buttons for the notification extension to stop notification calls. However, this may pose a security risk. Do not use these buttons if you do not really need them.

To find out what svn-halt buttons exist in the system, type **display svn-button-location** and press RETURN.

The SVN Button Locations screen appears.

```
SVN BUTTON LOCATIONS

LOGIN SECURITY VIOLATIONS

                            Name: Administrator______________
                       Extension: 81234____

REMOTE ACCESS SECURITY VIOLATIONS

                            Name: Administrator______________
                       Extension: 81234____

AUTHORIZATION CODE SECURITY VIOLATIONS

                            Name: Administrator II___________
                       Extension: 81235____

STATION SECURITY CODE VIOLATIONS

                            Name: Administrator II___________
                       Extension: 81235____
     ----------------------------------------------------------------------------
     --------------------------------------------------------------------------
```

SVN Referral Call With Announcement

The SVN Referral Call with Announcement option has the capacity to provide a recorded message identifying the type of violation accompanying the SVN referral call. Using Call Forwarding, Call Coverage, or Call Vector Time-of-Day Routing (to route to an extension or a number off the switch), SVN referral calls with announcements can terminate to a point on or off the switch.

Use of other means to route SVN referral calls to alternate destinations are not supported at this time. An attempt to use an alternate method to route SVN referral calls may result in a failure to receive the call or to hear the announcement.

Considerations

- You may only administer one referral destination per system for each type of violation.
- Exercise caution when administering bridged appearances for stations that are used as SVN referral destinations. SVN referral calls terminating to bridged appearances *must* be accompanied by an announcement message or *must* route to bridge appearances equipped with a display module. SVN referral calls that do not have an announcement and terminate to a bridged appearance not having a display will not provide an indication of the nature of the call.
- An authorization code violation with remote access generates two SVNs -- one displaying “authorization code violation” and one displaying “barrier code violation,” even though the correct barrier code was input. These two displays help you determine that the violation took place in the context of a remote access attempt, not an attempt to place an outgoing call to an ARS trunk.

Interactions

- Call Coverage, Call Forwarding, and Call Pickup

  These items are supported for SVN only if you use recorded announcements.
- Centralized Attendant Services (CAS)

  CAS attendants cannot receive referral calls from branch locations.
- Distributed Communications System (DCS)

  SVN does not support referral calls across a DCS network.

Related topics

‘‘Monitoring the Access Security Gateway history log’’ on page 390

Refer to ‘‘Login Administration’’ on page 970 to disable a login following a security violation.

Refer to ‘‘Setting up security violations notification’’ on page 395 for instructions.

Refer to ‘‘Recording announcements’’ on page 434 to record announcements.

Service observing

Designated users, normally supervisors, can listen to other users' calls. This capability is often used to train agents and monitor service quality in call centers and other environments where employees serve customers over the phone. On MultiVantage, this is called "service observing" and the user observing calls is the "observer."

This section describes service observing in environments without Automatic Call Distribution (ACD) or call vectoring. Refer to *Avaya MultiVantage™ Call Center Software Release 11 Guide to ACD Call Centers* to use service observing in those environments.

Brief description

Observers can monitor calls to any of the following:

- an extension
- a vector directory number (VDN) (on systems with call vectoring)
- a logical agent ID (on systems with Expert Agent Selection)

Note that service observing allows you to observe calls to one particular extension, not all calls to all extensions at a terminal.

Observers can monitor calls in one of two modes: "listen-only" or "listen-and-talk." The latter permits an observer to hear and speak with all parties on a call. The person being monitored doesn't know an observer is listening to the call unless you administer Avaya MultiVantage to provide a monitoring tone.

WARNING:
Listening to someone else's calls may be subject to federal, state, or local laws, rules, or regulations. It may require the consent of one or both of the parties on the call. Familiarize yourself with all applicable laws, rules, and regulations and comply with them when you use this feature.

Observers can use remote access to monitor calls when they're off-site. In systems with call vectoring, a vector can control access to service observing.

How to observe calls

Observers press the service observing button on their phone or dial a feature access code, and then dial the extension they want to observe. When using a service observing button, observers start in listen-only mode and can toggle between listen-only and listen/talk mode by pressing the button. The button lamp indicates which mode the observer is in.

If a service observer is in the wait state (waiting for an eligible call), the only visual indication the station/agent is observing is if the observing station has a Service Observe button administered. When this happens, the Service Observe button flutters while the observer is waiting, and the button is lit steady when the observer is actively observing a call.

If service observing is activated via the Service Observing feature access code, and there is no Service Observe button administered on the set, there will be no call appearance reserved while the call remains in the wait state. An idle call appearance must be available for the user to go to the observing state when a new eligible call arrives.

By contrast, there are different feature access codes for listen-only and listen-and-talk modes. When observers initiate sessions with a feature access code, they must choose one of the two modes at the start of the session. They cannot switch to the other mode without ending the session and beginning another. The feature access codes for service observing are:

- Service Observing Listen Only Access Code
- Service Observing Listen/Talk Access Code

NOTE:
Feature access codes are required for remote observing.

An observer can observe an agent who is not active on a call. The observer is in wait state until the agent receives a call, and then the observer is bridged onto the call.

To deactivate Service Observing, the observer hangs up, selects another call appearance, or presses the disconnect or release button.

Restrictions

Two observers can’t monitor the same extension or the same call simultaneously. If user A is observing an extension and user B tries to observe it, B gets a busy signal. If 2 extensions are being observed independently and one calls the other, only the observer of the calling extension observes the call. The observer of the called extension goes into wait state until the call is over.

Phone displays

A local observer’s phone display shows exactly what is displayed on the observed phone’s display, followed by the letters “so”.

Trunk calls

If a user makes a trunk-call, observation starts after the user finishes dialing. On central office (CO) trunks, dialing is considered complete when answer supervision is returned or when answer supervision timeout occurs.

Service observing cannot be activated over trunks without disconnect supervision. Any attempt is denied.

Warning and conference tones

If you administer a tone to notify the parties on a call when they are being observed, you can choose between a warning tone and conference tone. If you select warning tone, a unique 2-second, 440-Hz warning tone plays before an observer connects to the call. While the call is observed, a shorter version of this tone repeats every 12 seconds. If you select conference tone, all parties will hear conference tone before an observer connects to the call. However, unlike warning tone the conference tone is not repeated.

Interactions

- Attendants

  An attendant can be observed but cannot be an observer.

- Bridged Appearances

  You can only observe calls on primary extensions, not on bridged appearances. For example, let’s say you’re observing extension 3082 and this phone also has a bridged appearance for extension 3282. You can’t observe calls to 3282.

- Busy-Verification

  You can’t observe an extension that’s being busy-verified. You can’t busy-verify an extension that’s being observed.

- Call Coverage/Call Pickup

  An observer cannot observe a call answered by a covering agent or a member of a pickup group unless the called agent bridges onto the call.

- Call Park

  An observer cannot park the call they are observing.

- Call Waiting

  Incoming calls cannot wait on a single-line phone that is being observed.

- Conference

  Observers cannot initiate a conference while observing.

  If an observed user starts a conference or enters a conference with fewer than 6 parties, the observer is placed in wait state until the call is connected. Then the observer observes the conference and is counted as one party in the conference. (Conference members are observed during a conference regardless of their COR setting.) In addition, the observer is bridged onto any calls the user makes or receives before the conference is complete. When the user leaves the conference, the observer also leaves and returns to observing the original call.

- Data Privacy

  You can't observe an extension on which Data Privacy is active. You also can't observe an extension while it's on a conference call with another extension using Data Privacy.

- Data Restriction

  You can't observe an extension on which Data Restriction is active. You also can't observe an extension while it's on a conference call with another extension using Data Restriction.

- Integrated Directory

  Observers do not hear users dialing when the latter use this feature.

- DCS

  To observe stations on another node (a DCS station extension), you must set up remote-access service observing. Service observing displays are not transmitted across DCS networks.

- Hold

  Observers cannot place calls on hold while they're observing. If a user places a call on hold, the observer enters wait state.

- IP Solutions

  If you service observe into an ip-ip direct call, the people on the call may hear a break in conversation of about 200 ms.

- Leave Word Calling

  Parties on an observed call cannot use LWC.

- Music-on-Delay/Music-on-Hold

  If an observer is in listen/talk mode, neither caller nor observer hears music-on-hold. If an observer is in listen-only mode, the caller hears music-on-hold but the observer does not.

- Privacy — Manual Exclusion

  You can't observe an extension on which Privacy — Manual Exclusion is active. You also can't observe an extension while it's on a conference call with another extension using Privacy — Manual Exclusion.

- Transfer

  Observers cannot initiate a transfer while observing.

  If a user transfers a call, the observer is placed in wait state. The observer is bridged onto the call when the transfer is complete.

Related topics

Refer to ''Setting up automatic answer intercom calls'' on page 476 to administer service observing.

Single-Digit Dialing and Mixed Station Numbering

Single-Digit Dialing and Mixed Station Numbering allows easy guest access to internal hotel/motel services and provides the capability to associate room numbers with guest-room telephones.

Detailed description

You must create a dial plan for hotel/motel services and room numbers. (See ''Dial Plan Analysis Table'' on page 717 for more information about creating a dial plan.)

Some suggestions follow.

- Assign a single-digit extension to internal hotel/motel services such as room service. Assign single-digit extensions to individual-telephones or to a group of telephones (for example, to service the front desk).
- Assign a prefixed extension to guest rooms.

  A prefixed extension is still made up of a prefix and an extension up to **five** digits. The prefix identifies the call type. The switch collects dialed digits, removes the prefix digit, and uses the extension for further processing.

- Mixed station numbering extensions can have from one to seven digits and can begin with any digit from 0 to 9.

  The first digit, in combination with the number of digits dialed, defines the call type. To differentiate between two numbers with the same leading digit but different lengths, the system monitors the interval after a digit is dialed and before the next digit is dialed. If the interval extends past the administered interdigit timeout, the system assumes that dialing is complete and calculates the number of digits dialed up to that point.

Examples

```
change dialplan analysis                                        Page   1 of   3   SPE A
                             DIAL PLAN ANALYSIS TABLE
                                                                   Percent Full:    9


      Dialed  Total  Call        Dialed  Total  Call        Dialed  Total  Call
      String  Length Type        String  Length Type        String  Length Type
        0       1    attd
        1       2    dac
        2       1    ext
        3       1    ext
        4       1    ext
        5       3    ext
        6       1    ext
        6       4    pext
        7       1    ext
        7       5    pext
        8       1    dac
        9       1    dac
        *       2    fac
        #       2    fac
```

Dial plan 1 allows the following dial access:

- Single-digit access to the hotel/motel attendant (0)
- Ten dial-access codes (DACs) beginning with the digit 1 (10 through 19)
- Single-digit access to five hotel/motel services (2, 3, 4, 6 and 7)
- Nonprefixed access to as many as 100 hotel/motel staff extensions

  (500 through 599)
- Guest room extensions for as many as 100 floors
  - — Access to floors 1 through 9 (prefix digit 6 + [100 through 999])
  - — Access to floors 10 through 99 (prefix digit 7 + [1000 through 9999])
- Toll-call access via DAC 8

- Local calling via ARS DAC 9
- 2-digit feature-access codes (FACs) [* or # plus another digit

```
change dialplan analysis                                        Page   1 of   3   SPE A
                             DIAL PLAN ANALYSIS TABLE
                                                                    Percent Full:    9

      Dialed  Total  Call       Dialed  Total  Call       Dialed  Total  Call
      String  Length Type       String  Length Type       String  Length Type
        0       1    attd         5       2    ext
        1       1    ext          5       3    ext
        1       2    ext          6       1    ext
        1       3    ext          6       2    ext
        2       1    ext          6       3    ext
        2       2    ext          7       1    ext
        2       3    ext          7       2    ext
        3       1    ext          7       3    ext
        3       2    ext          8       1    dac
        3       3    ext          9       1    dac
        4       1    ext          *       2    fac
        4       2    ext          #       2    fac
        4       3    ext                       ext
        5       1    ext                       ext
```

Dial plan 2 allows the following dial access:

- Single-digit access to the hotel/motel attendant (0)
- Single-digit access to seven hotel/motel services (extensions 1 through 7)
- 2-digit access to 70 hotel/motel services (extensions 10 through 79)
- Guest-room extensions for floors 1 through 7 (extensions 100 through 799)
- Toll-calling access via DAC 8
- Toll-calling access via ARS feature access code 9
- 2-digit FACs (* or # plus another digit)

Cancel timeout intervals if the user dials # after dialing all required digits.

When using prefixed extension numbers, it is not necessary to include an entry for the "real" extension number in the dial plan. The server is able to complete a call using the prefixed extension number. When dialing 7345 (where 7 is the prefix), the communications server will ring extension 345.

When using a dial plan like the one above, which includes both prefixed and non-prefixed extensions, dialing 567 instead of 4567 will ring an administrative extension instead of a room.

The dialing delays, which may not be perceived by hotel guests, will occur when dialing 6 and 7. The server must wait for the 3- to 9-second interdigit timeout to expire before the call will be sent. The user can preempt the timer by pressing the # key after the number has been dialed.

NOTE:
When using prefixed extensions, the extension that shows up on a display phone does not show the prefix. The prefix will not show up on CDR reports. If extension number 3315 is prefixed with a 6 and the dial plan shows 3xxx for extensions, it is possible to dial either 3315 or 63315 to reach extension 3315. If the dial plan was changed to remove the entry for extensions in the 3xxx block, then 3315 could be reached only by dialing 63315.

Considerations

- Mixed Station Numbering allows guest room numbers and room extensions to be the same.
- You cannot assign prefixed extensions longer than seven digits (including prefix) to intercom lists.
- A trunk access code (TAC) and an extension can share a first digit only if the extension is shorter than the TAC.
- Although extensions with the same first digit can have different lengths, data-channel extensions must have the maximum number of digits to avoid timeout problems for data calls that the switch automatically sets up, for example, via the Call Detail Recording (CDR) link.
- An extension and a FAC can share the same first digit only if the extension is longer as long as they are not used for Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) faxes. These extensions work only within the switch; they do not work as remote uniform dialing plan (UDP) extensions.
- You should administer the Short Interdigit Timeout on the Feature-Related System Parameters screen to ensure that the delay between the end of dialing and call completion is not too long.

Interactions

- Attendant Display and Telephone Display

  Prefixed extensions display without the prefix. The return call button causes the prefix to dial, even though it does not display.
- Property Management System (PMS)

- PMS products accept only extensions of 5 digits or shorter. Therefore, a customer using PMS cannot use 6/7-digit extensions
- .Uniform Dial Plan

  The following limitations apply to a distributed communications system (DCS) environment:

  — Extensions that differ in length from the UDP do not distribute to other switches.

  — If the first two digits of an extension correspond to the floor number, floors cannot be serviced by more than one switch.

Station Hunting

Station Hunting routes calls made to a busy station down a chain of stations until one is found that is not active.

Detailed description

To use Station Hunting, you create a station hunting chain that governs the order in which a call routes from one station to the next when the called station is busy. Each station in the chain links to only one subsequent station. However, any number of stations may link to one station.

The system updates the calling party’s display with “h” when the system begins checking the station-hunting chain. Likewise, the system updates the display of the station that is hunted-to (the station that takes the call) with an “h.”

Calls route through the chain as follows.

Table 64. Station Hunting Characteristics

Condition	Response
Encounters an idle extension	Rings extension
Encounters an active extension	Routes to next extension in chain
Encounters an extension with a blank hunt-to station field	Returns busy tone if no station was idle
Encounters any station a second time	Returns busy tone
Has checked 30 stations in the chain, without finding an idle one	Returns busy tone

There is no limit to the number of extensions that can be in a station-hunting chain.

Station Hunting examples

In this example (Table 65), extension 2 is the called extension. Because extension 2 is busy, the system follows the station-hunting chain to find an idle extension. The system cannot find an idle extension so it returns busy tone to the caller. Note that the chain terminates with extension 5. This means that the system cannot route the call to extension 1 even though it is an idle extension in the chain.

Table 65. Station-Hunting Chain — Example 1

Extension	State	Rings on extension
1	Idle	2
2	Busy	3
3	Active	4
4	Active	5
5	Busy	

In this example (Table 66), extension 2 is the called extension. Because extension 2 is busy, the system follows the station-hunting chain to find an idle extension. The call is answered at extension 1.

Table 66. Station-Hunting Chain — Example 2

Extension	State	Rings on extension
1	Idle	2
2	Busy	3
3	Busy	4
4	Active	5
5	Active	1

In this example (Table 67), extension 2 is the called extension. Because extension 2 is busy, the system follows the station-hunting chain to find an idle extension. The system encounters extension 3 a second time without finding an idle station. The system stops checking the station-hunting chain and returns busy tone to the caller. Notice that both extensions 5 and 2 link to extension 3.

Table 67. Station-Hunting Chain — Example 3

Extension	State	Rings on extension
1	Idle	2
2	Busy	3
3	Busy	4
4	Busy	5
5	Busy	3

Station hunting options

You can administer the system to perform station hunting prior to sending calls to coverage. The Station Hunt Before Coverage option on the System-Parameters Call Coverage/Call Forwarding screen provides that when a call is made to a busy station, the switch checks to see if there is a hunt-to station assigned to the busy station. If there is, the switch tries to connect to the hunt-to station before going to coverage. If the hunt-to station is also busy, the switch continues hunting down the hunt-to chain. If all stations in the hunt-to chain are busy, the call goes to the dialed station's coverage.

Administration commands

When you remove a station, the system attempts to maintain a station-hunting chain. Consider the following examples:

- Station 1 links to 2 and 2 links to 3. If you remove station 2, the system links 1 to 3.
- Station 1 links to 2. Station 2 does not link to another extension. If you remove station 2, 1 no longer links to another extension.

When you duplicate a station, the extension in the hunt-to station field is not copied into the duplicated station.

When you execute "**list usage extension xxxxx**," the system displays all stations that contain the station's extension as their hunt-to station.

Interactions

Remember that the system checks the station-hunting chain only for idle and available extensions.

- Adjunct Switch Applications Interface

  The system attempts Station Hunting when ASAI routes to an extension with a hunt-to station.

- Automatic Call Distribution

  An agent extension can be part of a station-hunting chain. The system hunts the agent's chain only when the call is made directly to the agent's extension. Calls distributed through hunt groups to an ACD agent do not hunt the agent's station-hunting chain. Calls made to an extension for logical agents do not hunt the agent's station-hunting chain.

- Automatic Callback

  The system does not hunt the chain of the called extension when the call is a callback-return call.

- Bridged Appearance

  The system hunts the extension's station-hunting chain if the principal station has no call appearance at which the call can terminate, even though it may have available bridged appearances on other stations.

- Busy Verification

  The system does not attempt Station Hunting for busy-verify calls.

- Call Coverage

  Call Coverage has precedence over Station Hunting.

  Station Hunting is applied to the final coverage point following the final coverage point's hunt-to chain when the following conditions occur:

  — The Call Coverage screen's Hunt After Coverage field is **y**.

  — The last coverage point is unavailable (busy or no answer).

  — The last coverage point is a station with an assigned hunt-to station.

  — No one in the coverage path answered the call.

  Coverage — Don't Answer will cover the call after hunt if the call can terminate, but no one answers.

  If Station Hunt Before Coverage is active, a call to a busy station tries to terminate to the hunt-to phone before going to coverage. If the call does go to coverage, it is the coverage of the dialed extension (unless the phone is an XDID, and then the call goes to the coverage of the non-XDID phone found in the XDID's hunt-to field).

- Call Detail Recording

  CDR records the called extension, not the answering extension.

- Call Forwarding

  Call Forwarding has precedence over Station Hunting.

  If an idle station has Call Forwarding active, the system forwards the call. If a busy station has Call Forwarding active, a call to the station forwards. If the forwarded-to station is busy, the call follows that forwarded-to station's hunting chain.

  If the system finds Call Forwarding active at one of the stations in a station-hunting chain, it considers the station busy and bypasses it. The call goes to the next station in the chain.

- Call Park

  The system does not attempt Station Hunting on callpark-return calls.

- Call Pickup

  Call Pickup functions the same for calls terminating at a point in a station-hunting chain as it does for a regular calls.

- Call Vectoring

  You cannot assign a Vector Directory Number as a hunt-to station.

  If a **route-to** command's **with cov y** directs a call to a busy station, the call follows the station's hunt-to chain and not its coverage path. Refer to *Avaya MultiVantage™ Call Center Software Release 11 Call Vectoring and Expert Agent Selection (EAS) Guide* for more information.

- Call Waiting/Attendant Call Waiting

  Station Hunting has precedence over Call Waiting.

  If a called extension has Call Waiting active, and the extension is already busy on a call, the system hunts the station-hunting chain. If the system cannot terminate the call to a member of the chain, then the call waits at the called extension.

  If the system finds Call Waiting active at an extension in a station-hunting chain, it considers the extension busy and bypasses it.

- Class of Restriction

  The system checks the COR of the called extension; it does not check the COR of the hunt-to stations in the chain.

- Distributed Communications System

  Station Hunting is not a DCS feature. All members of a station-hunting chain must be on the same switch.

- Do Not Disturb

  When a phone has Do Not Disturb activated, a call to that phone goes to intercept treatment and not to station hunting.

- Extension Number Portability

  You cannot assign a remote ENP extension as a hunt-to station.

- Hunting/Hunting Group

  You cannot assign a direct departmental calling or Uniform Call Distribution extension as a hunt-to station.

- Intercom Call

  The system denies Station Hunting for intercom calls to a busy extension.

- Leave Word Calling

  If a caller initiates Leave Word Calling (LWC), the LWC message is left at the called extension even if the system uses Station Hunting in an attempt to complete the call.

- Multimedia

  Calls to multimedia endpoints must convert to voice before station hunting.

- Night Service

  The system denies Station Hunting when a night service call is made to a busy night-console extension.

- Outgoing Trunk Queueing

  The system does not attempt Station Hunting for an OTQ callback-return call.

- Personal Central Office Line

  The system does not attempt Station Hunting for a PCOL call.

- Personal Station Access

  The system considers a station with PSA dissociated as busy and bypasses it in the station-hunting chain.

- Priority Call

  The system denies Station Hunting for priority calls.

- Restriction

  The system applies proper intercept treatment to a restricted, called extension. However, the system *does not check restrictions on hunt-to stations*.

- Send All Calls

  Send All Calls coverage takes precedence over Station Hunting.

- Tenant Partitioning

  The system applies normal tenant restrictions to a call to the called extension. However, the system *does not check tenant restrictions on hunt-to stations*.

- Terminal Translation Initialization

  The system considers a station with TTI separation as busy and bypasses it in the station-hunting chain.

- Terminating Extension Group

  You cannot assign a TEG as a hunt-to station.

- Uniform Dial Plan

  You cannot assign a remote UDP extension as a hunt-to station.

- X-ported extension

  You can assign a hunt-to station to a station administered with X in the port field. It is treated as unavailable and skipped.

Related topics

Refer to ‘‘Station’’ on page 1066 for information to administer a Hunt-to-Station button.

Refer to ‘‘Coverage Path’’ on page 672 for information about station hunting after coverage.

Station Security Codes

Station Security Codes (SSC) provide security to you by preventing other users from accessing functions associated with your station. Each station user can change their own SSC if they know the station’s current settings.

You must create a system-wide SSC change feature access code (FAC) before users can change their SSC. You must also provide users with their individual SSC. A user cannot change a blank SSC.

Interactions

Users need a station security code to use the following system capabilities:

- Demand printing
- Extended User Administration of Redirected Calls
- Leave Word Calling
- Personal Station Access
- Voice Message Retrieval

Related topics

Refer to ''Assigning an extender password'' on page 360 for information about creating a station security code.

Refer to ''Training users'' on page 367 for information about changing your station security code.

Refer to ''Station'' on page 1066 for information about and field descriptions on the Station screen.

Refer to ''Security-Related System Parameters'' on page 1047 for information about and field descriptions on the Security-Related System Parameters screen.

Refer to ''Feature Access Code'' on page 752 for information about and field descriptions on the Feature Access Code screen.

Telephone Displays

Telephone displays provide multi-appearance phone users with current call and message information. The information that appears depends on the type of display the user selects with the buttons on the phone.

Retrieving stored information, such as messages and directory information, is easy and convenient. Users can select English, French, Italian, Spanish, or a user-defined language for their display.

With Enhanced Telephone Display, you can choose the types of characters that appear on user's phone displays. You can administer the switch to display standard Roman characters, or Cyrillic, Katakana, or Ukrainian characters. The character set displayed is determined by the phones your company uses.

Button display modes

You can assign several display modes to phone buttons. Users access these modes by pressing the assigned button on the phone. All the buttons are administrable.

Button Mode	Displays
Normal	Call-related information for the active call appearance, including the call appearance, calling- or called-party name and number, depending on the type of call. Elapsed Time can be invoked anytime the display is in normal mode. It displays elapsed time in hours, minutes, and seconds. Timing starts and stops when the button is pressed.
Inspect	Call-related information for an incoming call when the user is active on a different call appearance. You must reset the mode manually for each call.
Stored Number	One of the following numbers: ■ the last number that the user dialed (Last Number Dialed) ■ the number stored in an Abbreviated Dialing button administered to the phone ■ a number stored in an Abbreviated Dialing list ■ a number assigned to a button that was administered by Facility Busy Indication
Date and Time	Current date and time of day
Integrated Directory	Turns off the touch-tone signals and allows the user to use the touch-tone buttons to enter the name of a system user. After a name is entered, the display shows the name and extension. Integrated Directory can use 1 additional button: ■ Call-Disp - automatically returns the call requested by the currently-displayed message or the currently-displayed name and extension.

Button Mode	Displays
Message Retrieval	Retrieves messages for phone users. If no messages are stored, display shows NO MESSAGES. Messages can be retrieved even if the retriever is active on a call. Message Retrieval can use 3 additional related buttons: ■ Next Message - retrieves the next message or displays END OF FILE, PUSH Next TO REPEAT when in Retrieval mode. ■ Delete - deletes the currently displayed message. ■ Call-Disp - automatically returns the call requested by the currently-displayed message or the currently-displayed name and extension.
Coverage Message Retrieval Mode	Retrieves messages for phone users who do not have a display module assigned to their phone. You must administer retrieval permission for a user to be able to retrieve another user's messages. The retriever does not need to lift the handset to retrieve messages. Messages can be retrieved even if the retriever is active on a call. Coverage Message Retrieval can use 3 additional related buttons: ■ Next Message - retrieves the next message or displays END OF FILE, PUSH Next TO REPEAT when in Retrieval mode. ■ Delete - deletes the currently displayed message. ■ Call-Disp - automatically returns the call requested by the currently-displayed message or the currently-displayed name and extension.

Information on the display

Avaya MultiVantage provides the following call-related information:

- Call Appearance Identification

  The call appearance buttons are designated on the display by a lowercase letter. The display shows a= for a call incoming on the first button, b= for a call incoming on the second button, and so on.

  The system may omit the call-appearance information so that the Call Log find capability in the PC/PBX Connection software works properly.

- Calling Party Identification

  When a call is from inside the system, the display shows the caller's name or a unique identification administered for the phone being used, along with the calling party's extension. When the call is from outside the system, the display shows the trunk group name (such as CHICAGO) and the trunk access code assigned to the trunk group used for the call. If a user is active on a call and receives a subsequent call, the display automatically shows the identification of the subsequent caller for a few seconds, then automatically restores the display associated with the active call appearance.

  For example:

  Outgoing trunk call

```
b=87843541
```

  8 is the trunk access code and 784-3541 is the number dialed

  then

```
b=OUTSIDE CALL                 8
```

  or

```
b=WATS               101
```

  NOTE:
  Due to space limitations, some name displays are shortened to 15 characters. These include displays for transferred or covered calls, non-DCS ISDN-PRI call displays, vector directory number (VDN) service observing displays, and Leave Word Calling messages or the queue status of an agent.

- Called Party Identification

  On calls to a system user, the digits appear on the display as they are dialed. After dialing is complete, the called party's name and extension appears. If no name is accessed, the dialed digits remain on the display.

  On outgoing calls, the digits appear on the display as they are dialed. After dialing is complete, the display shows the name and trunk access code assigned to the trunk group being called. Optionally on a trunk-group basis, the display can show only the dialed digits, not the trunk group name and trunk access code.

For example:

dialed digits

```
a=3602
```

then

```
a=TOM BROWN             3062
```

or, if no name is available

```
a=EXT 3602              3062
```

- Call Purpose

  This identifies the reason for an incoming call or a redirected call. (A normal incoming call is not identified by a call purpose.) The following identifiers sometimes appear on the display:

Display	Meaning
b — (Busy)	The called user is active on a call, and has a temporary bridged appearance of the call.
c — (Cover All)	The called user has Cover All assigned.
callback	The call is an Automatic Callback call from the system.
d — (Coverage on Don't Answer)	The call was redirected because the called phone was not answered. Also indicates that the called user has a temporary bridged appearance of the call.
f — (Call Forwarding)	Another user has forwarded calls to this phone.
h — (Station hunt)	The called user is active on a call and station hunt was used to route the call.
ICOM	The call is an Intercom call.
p — (Pickup)	The user answered a Call Pickup group member's call.
park	The user parked a call.
priority	The call has priority status.
s — (Send All Calls)	The called user is temporarily sending all calls to coverage and the call has been redirected to this phone.

Message retrieval

Certain phones and the attendant groups can be designated for system-wide message retrieval. Users of these phones or consoles can retrieve Leave Word Calling (LWC) and Call Coverage messages for other phone users, including Direct Department Calling (DDC) groups, Uniform Call Distribution (UCD) groups, and Terminating Extension Groups (TEG). They can also retrieve external call logs. You can assign system-wide retrieving phones or consoles on the Feature-Related System Parameters screen.

Messages for a phone user can be retrieved at selected phones or any attendant console if the retriever is in the user's Call Coverage path and if permission to retrieve messages is assigned for the user's phone.

Enhanced Telephone Display

With Enhanced Telephone Display, you can choose the types of characters that appear on your phone displays. You can choose standard Roman characters, or Cyrillic, Katakana, or Ukrainian characters. Your Avaya representative sets the character type on the System Parameters Country-Options screen. The character set displayed is also determined by the phones your company uses.

You can choose one of the following character sets for messages on your display phones:

- Cyrillic contains the characters required to display the Russian language. All Russian characters appear in capital letters.
- Katakana contains the characters to display the Japanese language as well as some European characters and other symbols. All Japanese characters appear in capital letters.
- Roman contains two character sets:
  - US English contains the Roman alphabet, numerals, and special characters found on the standard US English keyboard. US English characters appear in capital and lowercase letters.
  - European contains characters for many European languages. All European characters appear in capital letters.
- Ukrainian contains the characters required to display the Ukrainian language. All Ukrainian characters appear in capital letters.

The type of phones your company uses must support the characters you want to display. Each character set requires specific firmware in the phone. Make sure you use phones with the same firmware type across your entire system, or the displays do not appear as expected. Your Avaya representative can make sure that you have the correct phone types for the characters you want to display.

Interactions

- DCS

  Trunk group and attendant information associated with a DCS call can be translated. If the displays are not associated with a DCS call, the name that appears is the name administered on the form used to administer the trunk group.

- Single-Digit Dialing and Mixed-Station Numbering

  If prefixed extensions are used in the system's dial plan, the prefix is not displayed when the extension is displayed. The Return Call button can be used to dial prefixed extensions, because the system dials the prefix, even though it is not displayed.

Interactions (Enhanced)

- Adjunct Switch Applications Interface (ASAI) and related adjuncts

  Information sent from the switch to any adjunct is the literal value of the field, not the enhanced characters. The display appears as a string of random characters — for example, as "2<@^."

- AUDIX Voice Power/Audix Voice Power Lodging

  Not supported.

- Data Call Setup

  Not supported.

- DCS

  All switches in a DCS network must have must have the same software load installed on each server, must have the enhanced characters enabled, and must have phones with the same firmware type.

- ECMA and QSIG Networking

  Information must be sent between Avaya MultiVantage systems.

- Leave Word Calling - Adjunct

  Not supported.

- Message Retrieval - Print Messages (Demand Print)

  Not supported.

- Monitor 1 and OneVision

  Monitor 1 and OneVision receive ASCII characters.

- OSSI

  OSSI displays the literal value of the display field, not the enhanced characters.

- Passageway Direct Connect

  Not supported.

- VUStats

  Use phones that support enhanced characters or screens may clear and displays will be incorrect.

Feature information displays

Phone displays provide information about the activity on individual phones and consoles, including confirmation that a certain feature is being used. The language used for messages is administered on the station screen for each phone.

This section shows the English, French, Italian, and Spanish message for each feature. When time is displayed, the English language uses AM and PM. All other languages use 24-hour time.

Table 68. Automatic Wakeup

English	French	Italian	Spanish
AUTO WAKEUP - Ext: xxxxx Time: --:-- xM	*REVEIL AUTO. - POSTE: xxxxx HEURE: --:--*	*SERVIZIO SVEGLIA - Tel: xxxxx Ora: --:--*	*DESPERT AUTOMA - EXT: xxxxx HORA: --:--*
INVALID EXTENSION - TRY AGAIN	*NUMERO DE POSTE EST ERRONE - REESSAYER*	*NUMERO ERRATO - RIPETERE*	*EXTENSION NO VALIDO - INTENTE DE NUEVO*
WAKEUP ENTRY DENIED - INTERVAL FULL	*DEM. REVEIL REFUSEE - INTERVALLE PLEIN*	*SVEGLIA NON ATTIVATA - ORARIO OCCUP*	*ENTRADA DENEGADA - INTERVALO COMPLETO*
WAKEUP ENTRY DENIED - NO PERMISSION	*DEM. REVEIL REFUSEE - SANS AUTORISATION*	*SVEGLIA NON ATTIVATA - NON PERMESSO*	*ENTRADA DENEGADA - SIN PERMISO*

Continued on next page

Table 68. Automatic Wakeup (Continued)

English	French	Italian	Spanish
WAKEUP ENTRY DENIED - SYSTEM FULL	*DEM. REVEIL REFUSEE - ENCOMBREMENT*	*SVEGLIA NON ATTIVATA - CONGESTIONE*	*ENTRADA DENEGADA - SISTEMA COMPLETO*
WAKEUP ENTRY DENIED - TOO SOON	*DEM. REVEIL REFUSEE - TROP TOT*	*SVEGLIA NON ATTIVATA - TROPPO PRESTO*	*ENTRADA DENEGADA - MUY PRONTO*
WAKEUP REQUEST CANCELED	*DEMANDE DE REVEIL EST ANNULEE*	*RICHIESTA SVEGLIA CANENTRYATA*	*SOLICITUD DE DESPERTADOR CANCELADA*
WAKEUP REQUEST CONFIRMED	*DEMANDE DE REVEIL EST CONFIRMEE*	*RICHIESTA SVEGLIA CONFERMATA*	*SOLICITUD DE DESPERTADOR CONFIRMADA*
Wakeup Call	*APPEL DE REVEIL*	*Serv. Sveglia*	*Despierte*

Table 69. ASAI

English	French	Italian	Spanish
You have adjunct messages	*MESSAGES SUPPLEMENTAIRES*	*MESSAGGI AGGIUNTIVI*	*TIENE MENSAJES ADICIONALES*

Table 70. Busy Verification of Terminals and Trunks

English	French	Italian	Spanish
ALL MADE BUSY	*TOUS OCC.*	*TUTTI OCCUPATI*	*TODAS OCUPADAS*
BRIDGED	*EN DERIVATION*	*OCCUPATO*	*PUENTEADA*
DENIED	*INTERDIT*	*NON PERMESSO*	*DENEGADO*
INVALID	*ERRONE*	*NON VALIDO*	*NO VALIDO*
NO MEMBER	*AUCUN MEMBRE*	*NESSUN ELEMENTO*	*NINGUN MIEMBRO*
OUT OF SERVICE	*HORS SERVICE*	*FUORI SERVIZIO*	*FUERA SERVICIO*
RESTRICTED	*RESTREINT*	*RISTRETTO*	*RESTRINGIDO*
TERMINATED	*TERMINE*	*TERMINATO*	*TERMINADO*
TRUNK SEIZED	*CIRCUIT SAISI*	*GIUNZIONE IMP.*	*ENLACE OCUPADO*
VERIFIED	*VERIFIE*	*VERIFICATO*	*VERIFICADO*

Table 71. Call Appearance

For each language, the active call appearance appears as:

"*a* =" (English)

Call-appearance buttons are shown on the display by a lower-case letter (a through z for the first 26 call appearances), followed by "=." Lower-case letters A through Z, followed by "=" are used for additional call appearances.

Table 72. Call Detail Recording

English	*French*	*Italian*	*Spanish*
CDR OVERLOAD"	*SURCHARGE EDA*	*SVRACCARICO DAC*	*SOBRECAR GA DAT*

Table 73. Call Progress Feedback Displays

English	French	Italian	Spanish
busy (Extension Busy, Intrusion Not Allowed, Call Waiting Not Allowed)	*OCCUPE* (Occupe)	*occ* (Occupato)	*OCUPADA* (Ocupada)
busy(I) (Extension Busy, Intrusion Allowed, Call Waiting Not Allowed)	*OCC.(E)* (Entree ligne occupe)	*occ(I)* (Occupato-Intrusione)	*OCUP(I)* (Ocupada-intrusion)
ringing (Extension Ringing)	*SONNE* (Libre)	*libero* (Libero)	*LIBRE* (Libero)
wait (Extension Busy, Intrusion Not Allowed, Call Waiting Allowed)	*ATTENTE* (Attente)	*auat* (Autoattesa)	*ESPERA* (Espera)
(I) wait (Extension Busy, Intrusion Allowed, Call Waiting Allowed)	*(E) ATTENTE* (Entree ligne attente)	*(I) auat* (Intrusione-Autoattesa)	*(I) ESPERA* (Intrusion, en espera)

Table 74. Class of Restriction Displays

Restriction	English	French	Italian	Spanish
Toll	TOLL	INT.	TASS	TARF
Full	FULL	COM.	DISB	LLEN
No Restrictions	NONE	AUC.	ABIL	NING
Origination	ORIG	DEP.	ORIG	ORIG
Outward	OTWD	SOR.	USCN	SALI

Use the following screens to translate time messages, if appropriate.

```
<DATE/TIME>     <TIME><b><b><DATE>
<TIME>          <HR>:<MIN><b><M>
<HR>            1-12 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<M>             "am" or "pm"
<DATE>          <DOW><b><MONTH><b><DOM>,<b><YEAR>
<DOW>           Day of week, upper case, unabbreviated
<MONTH>         Month of year, upper case, unabbreviated
<DOM>           1-31 (day of month, no leading zeroes)
<YEAR>          Year in 4 digits
<b>             Blank
```

Screen 279. Date/Time Mode and Formats – English

```
<DATE/TIME>     <TIME><b><b><DATE>
<TIME>          <HR>:<MIN>
<HR>            0-23 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<DATE>          <DOW><b><DOM><b><MONTH>,<b><YEAR>
<DOW>           Day of week, upper case, unabbreviated
<DOM>           1-31 (day of month, no leading zeroes)
<MONTH>         Month of year, upper case, unabbreviated
<YEAR>          Year in 4 digits
<b>             Blank
```

Screen 280. Date/Time Mode and Formats – French, Italian, Spanish, and User-Defined

Table 75. Days of the Week Format

English	French	Italian	Spanish
SUNDAY	DIMANCHE	DOMENICA	DOMINGO
MONDAY	LUNDI	LUNEDI	LUNES
TUESDAY	MARDI	MARTEDI	MARTES
WEDNESDAY	MERCREDI	MERCOLEDI	MIERCOLES
THURSDAY	JEUDI	GIOVEDI	JUEVES
FRIDAY	VENDREDI	VENERDI	VIERNES
SATURDAY	SAMEDI	SABATO	SABADO

Table 76. Date/Time Mode — Time Not Available

English	*French*	*Italian*	*Spanish*
SORRY, TIME UNAVAILABLE NOW	*HEURE ET DATE INDISPONIBLES*	*ORA E DATA TEMPON DISPONIBILI*	*HORA Y FECHA NO DISPONIBL ES AHORA*

Table 77. Months of the Year Format

English	French	Italian	Spanish
JANUARY	JANVIER	GENNAIO	ENERO
FEBRUARY	FEVRIER	FEBBRAIO	FEBRERO
MARCH	MARS	MARZO	MARZO
APRIL	AVRIL	APRILE	ABRIL
MAY	MAI	MAGGIO	MAYO
JUNE	JUIN	GIUGNO	JUNIO
JULY	JUILLET	LUGLIO	JULIO
AUGUST	AOUT	AGOSTO	AGOSTO
SEPTEMBER	SEPTEMBRE	SETTEMBRE	SEPTIEMBRE
OCTOBER	OCTOBRE	OTTOBRE	OCTUBRE
NOVEMBER	NOVEMBRE	NOVEMBRE	NOVIEMBRE
DECEMBER	DECEMBRE	DICEMBRE	DICIEMBRE

Table 78. Do Not Disturb (Hotel/Motel Feature)

English	*French*	*Italian*	*Spanish*
DO NOT DIST - Group: xx Time: --:-- xM	*NE PAS DERANGER GROUPE: xx HEURE: --:--*	*NON DISTURBARE - Grp: xx Ora: --:--*	*NO MOLESTAR - GRUPO: xx HORA: --:--*
DO NOT DIST - Ext: xxxxx Time: --:-- xM	*NE PAS DERANGER POSTE:xxxxx HEURE: --:--*	*NON DISTURBARE - Tel: xxxxx Ora: --:--*	*NO MOLESTAR - EXT: xxxxx HORA: --:--*
DO NOT DIST ENTRY DENIED - INTERVAL FULL	*DEMANDE EST REFUSEE - INTERVALLE PLEIN*	*SERVIZIO NON ATTIVATO - ORARIO OCCUP*	*ENTRADA DENEGADA - INTERVALO COMPLETO*
DO NOT DIST ENTRY DENIED - NO PERMISSION	*DEMANDE EST REFUSEE - SANS AUTORISATION*	*SERVIZIO NON ATTIVATO - NON PERMESSO*	*ENTRADA DENEGADA - SIN PERMISO*
DO NOT DIST ENTRY DENIED - SYSTEM FULL	*DEMANDE EST REFUSEE - ENCOMBREMENT*	*SERVIZIO NON ATTIVATO - CONGESTIONE*	*ENTRADA DENEGADA - SISTEMA COMPLETO*
DO NOT DIST ENTRY DENIED - TOO SOON	*DEMANDE EST REFUSEE - TROP TOT*	*SERVIZIO NON ATTIVATO - TROPPO PRESTO*	*ENTRADA DENEGADA - MUY PRONTO*
INVALID GROUP - TRY AGAIN	*GROUPE ERRONE - REESSAYER*	*GRUPPO NON VALIDO - RIPETERE*	*GRUPO NO VALIDO - INTENTE DE NUEVO*
THANK YOU - DO NOT DIST ENTRY CONFIRMED	*MERCI - DEMANDE EST CONFIRMEE*	*NON DISTURBARE - RICHIESTA CONFERMATA*	*NO MOLESTAR - ENTRADA CONFIRMADA*
THANK YOU - DO NOT DIST REQUEST CANCELED	*MERCI - DEMANDE EST ANNULEE*	*NON DISTURBARE - RICHIESTA CANENTRYATA*	*MUCHAS GRACIAS - SOLICITUD CANCELADA*

Enhanced Abbreviated Dialing -user defined

```
display display-messages ad-programming                        Page 1 of 1
                            Language Translations

     English

 1. Press button to program.         1. ****************************************
 2. Change number?                   2. ***********************
 3. Yes=1 No=2                       3. ****************
 4. Enter number:                    4. ****************************************
 5. Press # to save.                 5. ****************
 6. Number saved.                    6. ****************************************
 7. Change label?                    7. ***********************
 8. Enter label:                     8. ****************************************
 9. Press  * to advance; # to save.  9. ******************************
10. Press  * to reenter; # to save. 10. ******************************
11. Label saved.                    11. ****************************************
```

Table 79. Field Separator

English	French	Italian	Spanish
<calling party> *"to"* <called party>	<calling party> *"a"* <called party>	<calling party> *"a"* <called party>	<calling party> *"a"* <called party>

Table 80. Integrated Directory Display Model

English	*French*	*Italian*	*Spanish*
DIRECTORY - PLEASE ENTER NAME	*ANNUAIRE - ENTRER LE NOM*	*ELENCO UTENTI - INTRODURRE NOME*	*GUIA TELEFONICA - INTRODUZCA NOMBRE*
DIRECTORY UNAVAILABLE - TRY LATER	*ANNUAIRE INDISPONIBLE - REESSAYER*	*ELENCO UTENTI TEMP. NON DISPONIBILE*	*GUIA TEL INDISPONIBLE - INTENTE DESPUES*
NO MATCH - TRY AGAIN	*INTROUVABLE - REESSAYER*	*NESSUNA CORRISPONDENZA - RIPETERE*	*NO CORRESPONDE - INTENTE DE NUEVO*

Table 81. ISDN

English	*French*	*Italian*	*Spanish*
ANSWERED BY	*REPONDU PAR*	*RISPOSTA DA*	*RESPONDIDO POR*
CALL FROM	*APPEL DE*	*CHIAMATA DA*	LLAMADA DE
INTL	INTL	*INTL*	*INTL*

Leave Word Calling

```
<CALLER_ID><b><DATE><b><TIME><M><b><C><b>CALL<EXT_NO>

<CALLER_ID>     The calling identifier, up to 15 characters
<DATE>          <MONTH>/<DOM>
<MONTH>         1-12 (month of year, no leading zeroes)
<DOM>           1-31 (day of month, no leading zeroes)
<TIME>          <HR>:<MIN>
<HR>            1-12 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<M>             "a" or "p"
<C>             Number of calls received, 1 digit *
<EXT_NO>        Calling extension number, up to 7 digits
<b>             blank
```

Screen 281. Leave Word Calling Format – English

```
<CALLER_ID><b><DATE><b><TIME><b><C><b>APPL<b><EXT_NO>

<CALLER_ID>     The calling identifier, up to 15 characters
<DATE>          <DOM>/<MONTH>
<DOM>           1-31 (day of month, no leading zeroes)
<MONTH>         1-12 (month of year, no leading zeroes)
<TIME>          <HR>:<MIN>
<HR>            0-23 (hour of day, no leading zeroes)
<MIN>           00-59 (minute of hour)
<C>             Number of calls received, 1 digit *
<EXT_NO>        Calling extension number, up to 5 digits
```

Screen 282. Leave Word Calling Formats – French, Italian, Spanish, and User-Defined

Table 82. Leave Word Calling Messages

English	**French**	**Italian**	**Spanish**
CANNOT BE DELETED - CALL MESSAGE CENTER	*NE PEUT ETRE SUPP./APPELER RECEP. MESS.*	*NON CANENTRYATO. CHIAMARE CENTRO MESSAGGI*	*NO ELIMINADO-LLAMA CENTRO DE MENSAJES*
DELETED	*SUPPRIME*	*MESSAGGIO CANENTRYATO*	*ELIMINADO*
END OF MESSAGES (NEXT TO REPEAT)	*FIN DES MESSAGES (SUIVANT POUR REPETER)*	*FINE MESSAGGI. <successivo> PER RIPETERE*	*FIN DE MENSAJES (SIGUIENTE PARA REPITIR)*
GET DIAL TONE, PUSH Cover Msg Retrieval	*TONALITE D'ENVOI - <LECT. MESS. COUV.>*	*<rec mess copert> DOPO IL TONO DI CENTR*	*OBTENGA TONO OPRIMA <RECUP MNSJE COBERT>*
IN PROGRESS	*EN COURS*	*ATTENDERE...*	*EN CURSO*
MESSAGE RETRIEVAL DENIED	*LECTURE DE MESSAGES INTERDITE*	*LETTURA MESSAGGIO NON PERMESSA*	*RECUPERACION DE MENSAJES DENEGADA*
MESSAGE RETRIEVAL LOCKED	*LECTURE DE MESSAGES BLOQUEE*	*LETTURA MESSAGGIO BLOCCATA*	*RECUPERACION DE MENSAJES BLOQUEADA*
MESSAGES FOR	*MESSAGES POUR*	*MESSAGGI PER*	*MENSAJES PARA*
MESSAGES UNAVAILABLE - TRY LATER	*MESSAGES INDISPONIBLES - REESSAYER*	*MESSAGGI TEMPORANEAMENTE NON DISPONIBILI*	*MENSAJES NO DISPONIBLES, INTENTE DESPUES*

Continued on next page

Table 82. Leave Word Calling Messages (Continued)

English	**French**	**Italian**	**Spanish**
Message Center (AUDIX) CALL	*APPEL DE LA RECEPTION DE MESS. (AUDIX)*	*Chiamata dal Centro Messaggi (AUDIX)*	*LLAMADA DEL CENTRO DE MENSAJES (AUDIX)*
NO MESSAGES	*PAS DE MESSAGES*	*NESSUN MESSAGGIO*	*NINGUN MENSAJE*
WHOSE MESSAGES? (DIAL EXTENSION NUMBER)	*MESSAGES DE QUEL NO.? (ENTRER NO. POSTE)*	*LETTURA MESSAGGI. INTRODURRE NUMERO TEL.*	*MENSAJES DE QUIEN? (MARCAR EXTENSION)*

Table 83. Malicious Call Trace

English	**French**	**Italian**	**Spanish**
MALICIOUS CALL TRACE REQUEST	*DEPISTAGE D'APPELS MALVEILLANTS*	*RICHIESTA RINTRACCIO CHIAMATE MALEVOLE*	*RASTREO DE LLAMADA MALINTENCIO NADA*
MCT activated by: for:	*DAM ACTIVE par: pour:*	*RCM attivato da: per:*	*RLM activada por: para:*
original call redirected from:	*redirection appel initial de: (EXTENSION)*	*chiamata iniziale rinviata da:*	*llamada orig. transferida de:*
party: (EXTENSION)	*demandeur: (EXTENSION)*	*utente: (INTERNO)*	*usuario: (EXTENSION)*
party: (ISDN SID/CNI)	*demandeur: (NIP/INA ISDN)*	*utente: (NIC/INC ISDN)*	*usuario: (ISDN NIE/INU)*
party: (PORT ID)	*demandeur: (REF. PORT ISDN)*	*utente: (ID DELLA PORTA ISDN)*	*usuario: (ID DEL PUERTO ISDN)*

Continued on next page

Table 83. Malicious Call Trace

English	**French**	**Italian**	**Spanish**
party: (ISDN PORT ID)	*demandeur: (REF. PORT)*	*utente: (ID DELLA PORTA)*	*usuario: (ID DEL PUERTO)*
END OF TRACE INFORMATION	*FIN DES INFO DE DEPISTAGE*	*INFORMAZIONI FINALI SUL RINTRACCIO*	*FIN DE INFORMACION DE RASTREO*
voice recorder port:	*port enregistreur vocal:*	*porta del registratore:*	*puerto de grabado de voz:*

Table 84. Caller Information

English	***French***	***Italian***	***Spanish***
Info:	*INFO.:*	*Info:*	*INFORM:*

Table 85. Emergency Access to Attendant

English	***French***	***Italian***	***Spanish***
a=xxxxxxxxxxxxxxx Ext xxxxx xx in EMRG Q	*a=xxxxxxxxxxxxxx xx POSTE xxxxx xx FIL URG*	*a=xxxxxxxxxxxxxx xx Der xxxxx xx in C EMRG*	*a=xxxxxxxxx xxxxxx EXT xxxxx xx EN C EMRG*

Table 86. Queue Status

English	***French***	***Italian***	***Spanish***
HUNT GROUP <x> NOT ADMINISTERED	*GROUPE DE DIST. <x> NON ADMINISTRE*	*GRUPPO <x> NON AMMINISTRATO*	*GRUPO BUSQUEDA <x> NO ADMINISTRADO*

Table 87. Queue Status Indication

English	*French*	*Italian*	*Spanish*
<15 chrs> Q-time xx:xx calls xx	*<15 chrs> TEMPS-F xx:xx APPELS xx*	*<15 chrs> T-coda xx:xx chiam xx*	*<15 chrs> HORA-C xx:xx LLAMADAS xx*

Table 88. Miscellaneous Call Identifier

English	French	Italian	Spanish
sa (ACD Supervisor Assistance)	*AS* (Assistance surveillant)	*as* (Assistenza Supervisoree)	*AS* (Ayuda de supervisor)
ac (Attd Assistance Call)	*AA* (Appel assistance)	*ao* (Assistenza Operatore)	*AO* (Ayuda de operadora)
tc (Attd Control Of A Trunk Group)	*CF* (Commande faisceau)	*fc* (Fascio Controllato)	*CE* (Control enlaces)
an (Attd No Answer)	*TR* (Telephoniste sans reponse)	*on* (Operatore Non Risponde)	*ON* (Operadora no responde)
pc (Attd Personal Call)	*AP* (Appel personnel)	*cp* (Chiamata Personale)	*LP* (Llamada personal)
rc (Attd Recall Call)	*RA* (Rappel)	*rc* (Richiamata)	*RL* (Rellamada)
rt (Attd Return Call)	*RE* (Retour)	*rt* (Ritornata)	*RT* (Retorno)
sc (Attd Serial Call)	*AS* (Appel en serie)	*ic* (Inoltro a Catena)	*LS* (Llamada en serie)

Continued on next page

Table 88. Miscellaneous Call Identifier (Continued)

English	French	Italian	Spanish
co (Controlled Outward Restriction)	*RD* (Restriction de depart)	*cu* (Controllata Uscente)	*RS* (Restriccion saliente)
cs (Controlled Station to Station Restriction)	*RP* (Restriction vers postes)	*cd* (Controllata Derivati)	*CS* (Control estacion)
ct (Controlled Termination Restriction)	*AR* (Restriction d’arrivee)	*ct* (Controllata Terminante)	*RE* (Restriccion entrante)
db (DID Find Busy Station With CO Tones)	*OP* (Occupation du poste)	*po* (Passante Occupata)	*EO* (Estacion occupada)
da (DID Recall Go To Attd)	*RT* (Rappel telephoniste)	*pr* (Richiamata su Passante)	*RD* (Rellamada directa)
qf (Emerg. Queue Full Redirection)	*FP* (File d’urgence pleine deviation)	*de* (Deviata Emergenza)	*DE* (Desvio de emergencia)
hc (Held Call Timed Reminder)	*AG* (Indicatif d’appel en garde)	*at* (Avviso Chiamata in tenuta)	*LR* (Recordatorio de llamada retenida)
ic (Intercept)	*IN* (Interception)	*in* (Intercettata)	*IN* (Intercepcion)
ip (Interposition Call)	*AI* (Appel interposition)	*ip* (Interposizione)	*EP* (Entre posiciones)
ld (LDN Calls on DID Trunks)	*SD* (Selection directe)	*pd* (Diretta Passante)	*LD* (Larga distancia)

Continued on next page

Table 88. Miscellaneous Call Identifier (Continued)

English	French	Italian	Spanish
so (Service Observing)	*ES* (ecoute du service)	*is* (Inclusione Supervisore)	*SS* (Supervision del servicio)
na (Unanswered or Incomplete DID Call)	*SR* (Sans reponse)	*pn* (Passante Non Risposta)	*SR* (Sin respuesta)
ACB (Automatic Callback)	*R. AUTO.* (Rappel automatique)	*PRN* (Prenotazione Automatica)	*RA* (Rellamada automatica)
callback (Callback Call)	*RAPPEL* (Rappel)	*prenotaz* (Prenotazione)	*RELLAM* (Rellamada)
park (Call Park)	*G. I.* (garde par) indicatif	*parch.* (Parcheggiata)	*ESTAC* (Estacionamiento de llamada)
control (Control)	*CONTROLE* (Controle)	*cntr.op.* (Controllo Operatore)	*CONTROL* (Control)
ICOM (Intercom Call)	*INTERCOM* (Intercommunica tion)	*ICOM* (Intercom)	*INTERF* (Llamda interfono)
OTQ (Outgoing Trunk Queuing)	*FFD* (File faisceaux de depart)	*RFO* (Richiamata su Fascio Occupato)	*EES* (Espera de enlace de salida)
priority (Priority Call)	*PRIORITE* (Appel prioritaire)	*priorita* (Priorita’)	*PRIORIT* (Llamada prioritaria)
recall (Recall Call)	*APP.RAP.* (Appel rappel)	*richiam* (Richiamata)	*REPET* (Rellamada)
return (Return Call)	*RETOUR* (Retour)	*ritorno* (Chiamata Ritornata)	*RETORNO* (Llamada de retorno)

Continued on next page

Table 88. Miscellaneous Call Identifier (Continued)

English	French	Italian	Spanish
ARS (Automatic Route Selection)	*SAA* (Selection de l'acheminement automatiqe)	*SAI* (Selez. Autom. Instradam.)	*SAR* (Seleccion automatica de rutas)
forward (Call Forwarding)	*RENVOI* (Renvoi)	*deviata* (Deviata)	*REENVIO* (Reenvio de llamada)
cover (Cover)	*SUPPL.* (Suppleance)	*copert.* (Copertura)	*COBER* (Cobertura)
DND (Do Not Disturb)	*NPD* (Ne pas deranger)	*nd* (Non Disturbare)	*NM* (No molestar)
p (Call Pickup)	*P* (Prise)	*a* (Assente)	*C* (Captura de llamada)
c (Cover All Calls)	*s* (Suppleance)	*c* (Copertura)	*c* (Cobertura de toda llamada
n (Night Sta. Serv., Incoming No Answer)	*N* (Service nuit, entrant pas reponse)	*n* (Serv. Notte, Esterna Non Risposta)	*N* (Servicion noct. ext. no responde)
B (All Calls Busy)	*O* (Tous occupes)	*O* (Tutte Occupate)	*O* (Todas ocupadas)
f (Call Forwarding)	*R* (Renvoi)	*d* (Deviata)	*R* (Reenvio de llamada)

Continued on next page

Table 88. Miscellaneous Call Identifier (Continued)

English	French	Italian	Spanish
b (Cover Busy)	*o* (Suppleance occupee)	*o* (Copertura per Occupato)	*o* (Cobertura ocupada)
d (Cover Don't Answer)	*n* (Suppleance pas de reponse)	*n* (Copertura per Non Risposta)	*n* (Cobertura sin respuesta)
s (Send All Calls)	*E* (Envoi tous appels)	*r* (Rinvio)	*E* (Envio de toda llamada)

Table 89. User Identifiers

English	French	Italian	Spanish	Identifier
OPERATOR	TELEPHONISTE	OPERATORE	OPERADORA	Attendant
CONFERENCE	CONFERENCE	CONFERENZA	CONFERENCIA	Conference Call
EXT	POSTE	DER	EXTENSION	Extension
PAGING	PAGING	PAGING	PAGING	Paging (cannot be transplated)
OUTSIDE CALL	APPEL EXT.	ESTERNA	LLAMADA EXT.	Trunk Group
UNKNOWN NAME	INTROUVABLE	NOME SCONOSC.	DESCONOCIDO	Unknown

Table 90. Property Management System Interface

English	***French***	***Italian***	***Spanish***
CHECK IN - Ext:	*ENREGISTREMENT - POSTE:*	*CHECK IN - Tel:*	*REGISTRARSE - EXTENSION:*
CHECK IN: ROOM ALREADY OCCUPIED	*ENREGISTREMENT: CHAMBRE OCCUPEE*	*CHECK IN: CAMERA OCCUPATA*	*REGISTRARSE: HABITACION OCUPADA*
CHECK IN COMPLETE	*ENREGISTREMENT EFFECTUE*	*CHECK IN COMPLETATO*	*REGISTRO TERMINADO*
CHECK IN FAILED	*ECHEC D'ENREGISTREMENT*	*CHECK IN ERRATO*	*REGISTRARSE: FALLIDO*
CHECK OUT - Ext:	*DEPART - POSTE:*	*CHECK OUT - Tel:*	*PAGAR LA CUENTA - EXTENSION:*
CHECK OUT COMPLETE: MESSAGE LAMP OFF	*DEPART: PAS DE MESSAGES*	*CHECK OUT COMPLETATO: NESSUN MESSAGGIO*	*PAGO TERMINADO: NINGUN MENSAJE*
CHECK OUT COMPLETE: MESSAGE LAMP ON	*DEPART: MESSAGES*	*CHECK OUT COMPLETATO: MESSAGGI IN ATTESA*	*PAGO DE CUENTA TERMINADO: MENSAJES*
CHECK OUT FAILED	*ECHEC PROCEDURE DE DEPART*	*CHECK OUT ERRATO*	*PAGAR LA CUENTA: FALLIDO*
CHECK OUT: ROOM ALREADY VACANT	*DEPART - CHAMBRE INOCCUPEE*	*CHECK OUT: CAMERA NON OCCUPATA*	*PAGAR LA CUENTA: HABITACION VACANTE*
MESSAGE LAMP OFF	*PAS DE MESSAGES*	*NESSUN MESSAGGIO IN ATTESA*	*LUZ DE MENSAJE APAGADA*

Continued on next page

Table 90. Property Management System Interface (Continued)

English	***French***	***Italian***	***Spanish***
MESSAGE LAMP ON	*MESSAGES*	*MESSAGGI IN ATTESA*	*LUZ DE MENSAJE ENCENDIDA*
MESSAGE NOTIFICATION FAILED	*ECHEC D'AVIS MESSAGES*	*NOTIFICA MESSAGGI ERRATA*	*AVISO DE MENSAJE FALLIDO*
MESSAGE NOTIFICATION OFF - Ext: xxxxx	*AVIS DE MESSAGES DESACTIVE - POSTE:xxxxx*	*NOTIFICA MESSAGGI DISABIL. - Tel: xxxxx*	*AVISO DE MENSAJE APAGADO - EXT: xxxxx*
MESSAGE NOTIFICATION ON - Ext: xxxxx	*AVIS DE MESSAGES ACTIVE - POSTE:xxxxx*	*NOTIFICA MESSAGGI ABILITATA - Tel: xxxxx*	*AVISO DE MENSAJE ENCENDIDO - EXT: xxxxx*

Table 91. Security Violation Notification

English	***French***	***Italian***	***Spanish***
Barrier Code Violation	*VIOLATION DU CODE D'ENTREE*	*VIOLAZIONE DI CODICI DE TAGLIO*	*VIOLACIAON CONDIGO LIMITE*
Login Violation	*VIOLATION DE L'ACCES A L'ADMINISTRATION*	*IOLAZIONE DI INIZIO DI REGISTRAZIONE*	*VIOLACION CLAVE ACCESO*
Station Security Code Violation	*VIOLATION DE CODE D'ACCESS DE SECURITE*	*VIOLAZIONE DI CODICE DE SICUREZZA UTENTE*	*VIOLACION DE SEGURIDAD DE LA ESTACION*
Authorization Code Violation	*VIOLATION DEU CODE ACCES*	*VIOLAZION DEL CODICE D'AUTHORIZZAZIONE*	*VIOLACION DE CODIGO DE AUTORIZACION*

Table 92. Stored Number

English	*French*	*Italian*	*Spanish*
NO NUMBER STORED	*AUCUN NUMERO EN MEMOIRE*	*NESSUN NUMERO IN MEMORIA*	*NINGUN NUMERO ALMACENADO*

Table 93. Special Codes

English	French	Italian	Spanish
m (Mark)	*M* (Marquer)	*m* (Marcato)	*M* (Marca)
p (Pause)	*P* (Pause)	*p* (Pausa)	*P* (Pausa)
s (Suppress)	*S* (Supprimer)	*s* (Soppresso)	*S* (Suprimir)
w (Wait)	*A* (Attendre)	*a* (Attesa)	*E* (Espera)
W (Indefinite Wait)	*a* (Attendre)	*A* (Attesa)	*e* (Espera)

Station Hunting

Table 94. Calling Party Display

English	French	Italian	Spanish
HUNT	*Routage*	*Ricerca*	*Busqueda*

Table 95. Hunt-to Station Display

English	French	Italian	Spanish
h	*r*	*r*	*b*

In the following displays, x and y denote the Route Plan Number (RPN 1-8), yyy is a 3-letter abbreviation for the day of the week, and zz:zz is the activation time (24-hour time). Also below is the table that lists the 3-letter abbreviations for the day of the week.

Table 96. Time-of-Day Routing Messages

English	*French*	*Italian*	*Spanish*
ENTER ACTIVATION ROUTE PLAN, DAY & TIME	*ENTRER PLAN D'ACTIVATION, JOUR ET HEURE*	*INTRODURRE PIANO DA ATTIV., GIORNO E ORA*	*INTRODUZCA PLAN ACT DE RUTAS, DIA Y HORA*
ENTER DEACTIVATION DAY AND TIME	*ENTRER JOUR ET HEURE DE DESACTIVATION*	*INTRODURRE GIORNO E ORA DI DISATTIVAZ*	*INTRODUZCA DIA Y HORA DE DESACTIVACION*
OLD ROUTE PLAN: x ENTER NEW PLAN:	*ACHEMINEMENT ANT.: x ENTRER NOUVEAU:*	*INSTRADAMENTO PREC: x INTROD IL NUOVO:*	*PLAN RUTAS ANT: x INTRODUZCA EL NUEVO:*
OLD ROUTE PLAN: x NEW PLAN: y	*ACHEMINEMENT ANT.: x NOUVEAU PLAN: y*	INSTRADAMENTO PREC: x NUOVO PIANO: y	*PLAN RUTAS ANT: x NUEVO PLAN: y*
ROUTE PLAN: x FOR yyy ACT-TIME: zz:zz	*ACHEM.: x POUR yyy ACT-HEURE: zz:zz*	*INSTRADAMENTO: x PER yyy ATTIV ORE:zz:zz*	*PLAN RUTAS: x PARA yyy HORA-ACT: zz:zz*
ROUTE PLAN: x FOR yyy DEACT-TIME: zz:zz	*ACHEM.: x POUR yyy DESACT-HEURE: zz:zz*	*INSTRADAM.: x PER yyy DISATTIV ORE:zz:zz*	*PLAN RUTAS: x PARA yyy HORA-DESACT:zz:zz*

To enter the day of the week, the user dials 1 for Sunday, 2 for Monday, and so on.

Table 97. Time-of-Day Routing Days of the Week

English	French	Italian	Spanish
Mon	*LUN*	*Lun*	*LUN*
Tue	*MAR*	*Mar*	*MAR*
Wed	*MER*	*Mer*	*MIE*
Thu	*JEU*	*Gio*	*JUE*
Fri	*VEN*	*Ven*	*VIE*
Sat	*SAM*	*Sab*	*SAB*
Sun	*DIM*	*Dom*	*DOM*

Table 98. Transfer Messages

English	French	Italian	Spanish
TRANSFER COMPLETED	TRANSFERT EFFECTU	TRASFERIMEN TO COMPLETATO	TRANSFEREN CIA REALIZADA

Mapping enhanced display characters

Use the tables below to map US English characters to Russian, Japanese, European, or Ukrainian characters. Characters appear on the display terminal in the order in which you enter them. If you want the display to read right to left, enter the characters in reverse order on the screen.

US English to Russian characters

Table 99. US English to Russian characters

Russian	US English	Russian	US English
space	space	…	Q
‘	A	þ	R
»	B	¤	S
—	C	Ý	T
¬	D	Ð	U
”	E	Ã	V
¿	F	÷	W
÷	G	Þ	X
–	H	Õ	Y
ÿ	I	?	Z
Œ	J	X	{
À	K	⁄	}
ƒ	L	ý	:
‹	M	›	“
“	N	¡	<
Ÿ	O	?	>
«	P		

US English to Japanese characters

Table 100. US English to Japanese characters

Japanese	US English	Japanese	US English
space	space	ｸ	8
｡	!	ｹ	9
｢	“	ｺ	:
｣	#	ｻ	;

Continued on next page

Table 100. US English to Japanese characters (Continued)

Japanese	US English	Japanese	US English
、	$	シ	<
•	%	ス	=
ヲ	&	セ	>
ァ	‘	ソ	?
ィ	(	タ	@
ゥ	)	チ	A
ェ	*	ツ	B
ォ	+	テ	C
ャ	,	ト	D
ュ	-	ナ	E
ョ	.	ニ	F
ッ	/	ヌ	G
ー	0	ネ	H
ア	1	ノ	I
イ	2	ハ	J
ウ	3	ヒ	K
エ	4	フ	L
オ	5	ヘ	M
カ	6	ホ	N
キ	7	マ	O
ミ	P	ρ	f
ム	Q	g	g
メ	R	√	h
モ	S	−1	i
ヤ	T	j	j
ユ	U	∗	k
ヨ	V	¢	l

Continued on next page

Table 100. US English to Japanese characters (Continued)

Japanese	US English	Japanese	US English
ﾗ	W	£	m
ﾘ	X	ñ	n
ﾙ	Y	ö	o
ﾚ	Z	p	p
ﾛ	[	q	q
ﾜ	\	θ	r
ﾝ	]	∞	s
ﾞ	^	Ω	t
ﾟ	_	ü	u
α	‘	Σ	v
ä	a	π	w
β	b	$\bar{x}$	x
μ	d	y	y
σ	e	÷	}

For Japanese, the z, {, and | characters map to Kanji characters as follows:

z—symbol for 1,000

{—symbol for 10,000

|—symbol for Yen

US English to European characters

Some of the characters in the following map appear in only upper- or lower-case — for example, , Ê, ø, and others.

Table 101. US English to European characters

European	US English	European	US English
Ì	space	»	8
Ô	!	Ë	9
„	“	£	:
·	#	š	;
‡	$	›	<
	%	¯	=
	&	Ê	>
È	‘	”	?
Ë	(	Ó	@
Ê	)	Â	A
Š	*	¼	B
œ	+	¸	C
˝	,	˚	D
z̊	-		E
	.	Í	F
Ú	/	³	G
Õ	0	Í	H
œ	1	«	I
Đ	2	¯	J
¡	3	–	K
¿	4	ÿ	L
⁄	5	Ò	M
Ÿ	6	ý	N

Continued on next page

Table 101. US English to European characters (Continued)

European	US English
…	7
Œ	P
Ý	Q
Ð	R
¬	S
¤	T
Ÿ	U
þ	V
ð	W
þ	X
å	Y
ÿ	Z
¥	[
Ÿ	\
—	]
—	^
‘	_
Ï	‘
‰	a
?	b
°	d
‹	e
Ï	f
Î	g
	O
š	h
ä	i
?	j
ø	k
z¥	l
Ú	m
Ò	n
ˆ	o
Ã	p
ƒ	q
„	r
Ð	s
¤	t
¸	u
Ã	v
À	w
–	x
½	y
ç	z
	{
ø	|
“	}

US English to Ukrainian characters

Table 102. US English to Ukrainian characters

Ukrainian	US English	Ukrainian	US English
space	space	…	Q
‘	A	þ	R
»	B	¶	S
—	C	Ý	T
¬	D	Ð	U
”	E	Ã	V
¿	F	÷	W
÷	G	Þ	X
–	H	Õ	Y
ÿ	I	?	Z
Œ	J	X	{
À	K	ð	}
ƒ	L	ý	:
‹	M	™	“
“	N	¡	<
Ÿ	O	?	>
«	P		

Related Topics

Refer to System Parameters Country-Options for more information about and field descriptions on the System Parameters Country-Option screen.

Refer to Changing the display language for information about displaying the Russian, Japanese, European, and Ukrainian character sets.

Temporary Bridged Appearance

Temporary Bridged Appearance allows multiappearance telephone users in a terminating extension group (TEG) or personal central office line (PCOL) group to bridge onto an existing group call. If a call has been answered using the Call Pickup feature, the originally called party can bridge onto the call. This feature also allows a called party to bridge onto a call that redirects to coverage before the called party can answer it.

Detailed description

An incoming call to a TEG or PCOL group is not a call to an individual, although one particular member of the group can be the most qualified person to handle the given call. If this individual did not answer the call originally, this individual can bridge onto the call; the call does not have to be transferred.

A call to an individual can be answered by a call pickup group member. If the called party returns while the call is still connected, the called party bridges onto the call and the answering party hangs up.

Call Coverage provides redirection of calls to alternate answering positions (covering users). A temporary bridged appearance is maintained at the called telephone.

The called party can answer the call at any time, even if it is already answered by a covering user. If the called party does not bridge onto the call, the covering user can use the Consult function of Call Coverage to determine if the called party wants to accept the call. The Consult function uses the temporary bridged appearance maintained on the call. When the consult call is finished, the temporary bridged appearance is removed.

Stations that normally have a temporary bridged appearance with their coverage point do not have a temporary bridged appearance if the coverage point is AUDIX.

Keep Held SBA at Coverage Point

Considerations

- Temporary Bridged Appearance allows the desired party to bridge onto a call without manually transferring the call, providing convenience of operation and saving time.
- Temporary Bridged Appearance does not provide any call originating capability or the capability to answer another party's calls. These capabilities are provided by the Bridged Call Appearance feature.
- If two parties are bridged together on an active call with a third party, and if the Conference Tone feature is enabled, conference tone is heard.
- The Bridged Call Appearance feature enhances Temporary Bridged Appearance by allowing more than one call to an extension to be bridged and by allowing calls to be originated from bridged appearances.

Interactions

- Call Coverage

  Calls redirected to Call Coverage maintain a temporary bridged appearance on the called telephone if a call appearance is available to handle the call. The called party can bridge onto the call at any time. The system can be administered to allow a temporary bridged appearance of the call to either remain at or be removed from the covering telephone after the principal bridges onto the call. If two parties are bridged together on an active call with a third party, and the bridging tone is administered to yes, all three parties hear the bridging tone.

- Consult

  Consult calls use the temporary bridged appearance maintained on the call. At the conclusion of a consult call, the bridged appearance is no longer maintained. If the principal chooses not to talk with the calling party, the principal cannot bridge onto the call later.

- Conference and Transfer

  If a call has, or has had, a temporary bridged appearance; is conferenced or transferred; and redirects to coverage again; a temporary bridged appearance is not maintained at the conferenced-to or transferred-to extension.

- Privacy — Manual Exclusion

  When Privacy — Manual Exclusion is activated, other users are prevented from bridging onto a call. A user who attempts to bridge onto a call when this feature is active is dropped.

Tenant Partitioning

(Not available with Offer B) Tenant Partitioning provides telecommunications services to multiple independent groups of users through a single Avaya MultiVantage. Most commonly, Tenant Partitioning provides these services from a single provider to multiple tenants of an office complex. This eliminates the need for each tenant to purchase services separately, while still giving each tenant the appearance of a dedicated Avaya MultiVantage. You can also use this feature to provide group services, such as departmental attendants, on a single-customer Avaya MultiVantage. Tenant Partitioning also allows you to assign a unique music source for each tenant partition for callers who are put on hold.

NOTE:
If you use equipment that rebroadcasts music or other copyrighted materials, you may be required to obtain a copyright license from, or pay fees to, a third party such as the American Society of Composers, Artists, and Producers (ASCAP) or Broadcast Music Incorporated (BMI). You can purchase a Magic-on-Hold® system, which does not require such a license.

Detailed description

Tenants are defined and assigned by you, the system administrator. You must have the same authorization as that required for Class of Restriction (COR) administration.

Because some features are not partitioned, you must take care to administer these features to prevent inter-tenant access. Refer to ''Interactions'' on page 1755 for a list of these features.

You must ensure that:

- All tenants can call and be called by partition 1. This is the system default. If you change this default some call types fail. For example, *dial 0 fails*, as do SVN calls, ACA calls, etc.
- All stations in a call-pickup group are under control of the same tenant
- All stations with bridged appearances are under control of the same tenant
- Stations in different departments (for the purposes of attendant services) can call each other

You must assign a tenant partition number to each object (endpoint, virtual endpoint, or other entity) that has an assigned COR. The exceptions are authorization codes and fixed-assignment virtual endpoints.

You administer Tenant Partitioning via the Tenant Partitioning screen; you need to administer one screen for each tenant partition. Begin the initial administration of the Tenant Partitioning feature by completing the tenant partitioning screen. Keep in mind that you must specify an attendant group for each tenant that you define, even if there are no consoles assigned to the attendant group. You must also assign an attendant console to a tenant partition and you must assign a group number to the Attendant Console screen.

Tenant Partitioning capabilities

Tenant Partitioning can provide the following services to tenants: telephone equipment, building wiring, public and private network access, and attendant services. In addition, the feature can provide a full range of Avaya MultiVantage capabilities to even the smallest tenant office, including Call Coverage, Remote Access, Night Service Routing, and others. Tenants can also purchase Avaya MultiVantage adjuncts available on the switch, such as Call Management System (CMS) activity reporting.

Tenant Partitioning provides advantages to both the telecommunications service provider and individual tenants:

- Shared resources offer enhanced services at lower cost to the tenant, with increased profit for the service provider.
- The tenant has the appearance of a dedicated Avaya MultiVantage without the expense.
- All tenants can have attendant services.
- A trained, full-time system administrator can install, administer, and maintain the switch.

With proper administration, tenant resources, including trunking facilities, and all other switch endpoints can be protected from access by other tenants.

Partitioning tenants

The default for Tenant Partitioning is one universal tenant for the system. This tenant, partition 1, is usually reserved for the service provider. By default it has access to all facilities and any other tenant can access it.

The service provider creates additional partitions based on tenant requirements. When deciding which tenant partitions to create, remember:

- You can assign each switch endpoint to one and only one tenant partition. And, you must pass each switch endpoint to a partition. For example, you must assign each telephone, attendant console, trunk, and virtual endpoint, such as an listed directory number (LDN) or vector directory number (VDN), to a tenant partition.

- Most tenant partitions are discrete, separate units. By default, the system prevents all tenants, except partition 1, from accessing stations or trunking facilities belonging to other tenants. However, you can change this default. You can give explicit permission for one tenant to access another. For example, you can allow tenant 6 to call tenants 9 and 16 only.

  NOTE:
  If a tenant has permission to call another tenant, it has access to every endpoint belonging to that tenant. For example, if tenant 6 has permission to call tenant 9, tenant 6 can also use any trunking facilities present in tenant partition 9.

- Even if two extensions are blocked from calling each other by Tenant Partitioning restrictions, either extension can still reach the other by dialing the extension's Direct Inward Dialing (DID) number via the public network.
- If any facilities are to be shared among tenants who do not want complete access to each other's facilities, you must group the shared facilities in a separate partition. For example, if two tenants share a trunk, but do not have direct access to each other's telephones, that trunk will need to be placed in its own partition so that both tenants can access it.

It is also important to consider the following constraints and requirements of access control, attendant services, music sources on hold, and network route selection when you establish or assign partitions.

Access control

Features such as call coverage are limited by tenant-to-tenant access restrictions. For example, suppose tenant 1 includes a telephone from tenant 2 in its coverage path. If tenant 3 has permission to call tenant 1 but not tenant 2, a call from tenant 3 to tenant 1 skips the tenant 2 coverage point.

You may also want to set up tenants with special access privileges. For example, you might give a restaurant in an office complex permission to be called by any other tenant. Likewise, permission to call or be called by other tenants is helpful for building security or Avaya MultiVantage administration or troubleshooting.

You can also assign all CO trunks to one tenant partition that can then be accessed by all other tenants.

Attendant services

Tenant Partitioning allows you to provide personalized attendant services to each tenant.

The system provides one principal and one night or day/night attendant per attendant group. You assign each tenant an attendant group for service. Each attendant group has a separate queue. Queue warning lamps remain dark when Tenant Partitioning is active. However, information displayed when someone presses a queue-status button reflects the status of the attendant-group queue. The total number of calls queued for all tenants cannot exceed the system limit.

Attendant groups may serve more than one tenant. In this case, the attendant group cannot extend a call from one tenant via facilities belonging to another tenant, unless the former tenant has permission to access the other’s facilities.

Each tenant may have a designated night-service station. The system directs calls to an attendant group in night service to the night-service station of the appropriate tenant (when a night attendant is not available). When someone places an attendant group into night service, all trunk groups and hunt groups that belong to tenants served by that attendant group go into night service. In this case, the system routes incoming calls to the night-service destination of the appropriate tenant. Each tenant can have its own LDN night destination, trunk answer on any station (TAAS) port, or night attendant.

An attendant can specify that access to a trunk group is under attendant control if the trunk group is assigned to a tenant served by that attendant’s group. The system directs any valid user attempt to access the trunk group to the attendant group serving the tenant.

Network route selection

You can place trunk groups belonging to different tenants in the same route pattern. Calls routing to that pattern select the first trunk group in the pattern with access permission by the calling tenant (subject to normal constraints).

Tenant partitioning examples

The following is a simple example of how you might administer Tenant Partitioning in an office complex.

You assign tenant partition 1, the universal tenant, as the service provider. All other tenants can call and be called by the service provider.

You assign tenant partitions 2–15 to individual businesses in the complex. You maintain the system-default restrictions for these tenants. That is, tenants cannot access telephones, trunking facilities, or other switch endpoints belonging to other tenants.

You assign tenant partition 16 to the restaurant in the building complex. You give all tenants permission to call this tenant. However, to prevent the restaurant from accessing trunks and other facilities belonging to tenants, you do not permit the restaurant to call any other tenants.

You assign tenant partition 17 to all Central Office (CO) trunk groups. You give all tenants permission to call this tenant.

You assign tenant partition 18 to a trunk group that tenants 3 and 7 want to share. You give Tenants 3 and 7 access to this partition; you deny all other tenants access. To prevent toll fraud, you do not allow tenant 18 to call itself.

The Automatic Route Selection (ARS) route pattern can be the same for all tenants. In this example, the trunk for tenant partition 18 (the private trunk shared by tenants 3 and 7) is first in the route pattern. Tenant partition 17 is second. Tenants 3 and 7 route first to partition 18 and then as a second choice to partition 17. You deny all other tenants access to partition 18 and so the system routes them directly to partition 17.

All facilities that are not shared, including trunk groups, VDNs, telephones, attendant consoles, and other endpoints, are assigned to the tenant partition that they serve.

Table 103 on page 1753 summarizes the calling permissions for the different partitions. Yes indicates that the partitions have permission to call and be called by each other; no indicates that partitions cannot call or be called by each other.

Table 103. Calling permissions for partitions

Calling tenant partition number	Called tenant partition number					
	1	**2, 4—6, 8—15**	**3,7**	**16**	**17**	**18**
1	yes	yes	yes	yes	yes	yes
2, 4-6, 8-15	yes	Each partition can call itself but not the others	no	yes	yes	no
3,7	yes	no	Each partition can call itself but not the others	yes	yes	yes

Continued on next page

Table 103. Calling permissions for partitions (Continued)

Calling tenant partition number	Called tenant partition number					
	1	**2, 4—6, 8—15**	**3,7**	**16**	**17**	**18**
16	yes	no	no	yes	yes	no
17	yes	yes	yes	yes	yes	no
18	yes	no	yes	no	no	no

Detailed description of Multiple Music-on-Hold

Tenant Partitioning allows you to assign each tenant a music source, unique to each tenant partition, to be heard when a call is placed on hold. The tenant number assigned to the destination extension usually determines which music source is heard. This capability allows you to tailor the music or messages for the business needs of each tenant partition.

If the COR of the extension that places the call on hold permits music-on-hold, a caller on hold hears the music source assigned to the partition at which the call initially terminates. For example, if calls coming into the Avaya MultiVantage route first to an INTUITY automated attendant that then routes the call to the appropriate tenant partition, the caller on hold hears the music source of the INTUITY automated attendant, not the tenant partition to which it is routed. Likewise, if a caller in tenant partition 2 makes an out-going call using tenant partition 1’s trunk groups, the caller will hear the music source assigned to tenant partition 1. If the COR of the called extension does not permit music on hold, however, the caller hears nothing.

The maximum number of music sources allowed is the same as the maximum number of tenant partitions allowed; each music source can be used by one or more tenant partitions.

You can assign one of the following music-on-hold types to each tenant partition.

Table 104. Music-on-Hold Types

Type	System Response for a caller placed on hold
none	silence
tone	system-wide administered tone
music	the music associated with the administered port. The number of possible music sources equals the number of possible tenant partitions. Each partition can have its own music source.

Interactions

Tenant-partition identification is not passed between switches. A network of Avaya MultiVantage systems does not enforce Tenant Partitioning restrictions without special administration. For example, Tenant Partitioning on a network of Avaya MultiVantage systems does not enforce tenant-specific tie trunks.

Administration of the following features requires special care to avoid undesired intertenant access.

- Bridging
- Call Pickup
- Call Vectoring
- Controlled Restriction
- Facility Busy Indication
- Facility Test Calls
- Integrated Directory
- Inter-PBX Attendant Calls
- Main/Satellite/Tributary
- Malicious Call Trace
- Personal CO line
- Private Networking (AAR)
- Service Observing
- Uniform Dial Plan

The function of any feature that specifies a tenant partition is affected by tenant-to-tenant restrictions, as follows.

- AAR/ARS

  Do not confuse tenant partitions with Time-of-Day Plan Numbers and Partition Groups in AAR/ARS. You can still use Time-of-Day Plan Numbers and Partition Groups can still be used to select one of eight route patterns for AAR/ARS routing when Tenant Partitioning is in effect.

- Attendant and Attendant Group Features

  Tenant Partitioning creates multiple attendant groups. Attendant operations such as direct-station or trunk-group select (DCS/DTGS) are subject to tenant-to-tenant restrictions, both at selection time and at split time.

  All calls put on hold by an attendant from the attendant group hear the music source from the attendant group.

- Attendant Control of Trunk-Group Access

  An attendant group controls access only to trunk groups that belong to tenants that are served by that attendant group.

- AUDIX, embedded AUDIX, and AUDIX Voice Power

  AUDIX voice and data ports are subject to the same tenant-to-tenant restrictions as any other endpoint.

  AUDIX can restrict one group of subscribers from sending voice mail to another group. The tenant-partitioning provider can create up to 10 different communities within each AUDIX that either have or do not have permission to send voice messages across community boundaries.

- Authorization Codes

  Authorization codes are associated with classes of restriction. If you want to have a set of authorization codes that is unique to a given tenant, you could select a group of CORs for a that tenant and only assign those CORs to objects in that partition.

- Automatic Wakeup

  Wakeup music will be the music source assigned to the wakeup station's tenant partition.

- Bridged Call Appearance

  All stations with bridged call appearances should be administered to be under control of the same tenant.

- Call Coverage

  Tenant-to-Tenant access restrictions apply to coverage paths. If a tenant cannot access a particular tenant, it cannot access that tenant as part of another tenant's coverage point.

  When an attendant is specified as part of a coverage path, the attendant group of the called tenant, not the calling tenant, is accessed.

  When a call goes to coverage, is answered, and is put on hold, the music on hold is the music source assigned to the tenant partition of the terminal that was originally called.

- Call Detail Record (CDR)

  CDR does not report the tenant partition number of the extension or trunk group used. You must infer the tenant partition number from the extension or trunk-group number.

- Call Pickup

  Administer all stations in a call-pickup group to be under control of the same tenant. The system supports Call Pickup only if the caller and the called party can both call the pickup user. The caller and the called party do not need to be in the same pickup group.

- Call Vectoring/VDN

  A caller routed to a new destination by a vector step hears the music assigned to the last active VDN. While a call is in vector processing, the tenant number (TN) assigned to the active VDN (as determined by VDN Override) determines the music source heard by callers on hold in most circumstances. Note the following exception, however.

  If you use a *wait-time <time> hearing <extension> then <treatment 2>* command where the *<extension>* is a music source (assigned on the Announcements/Audios Sources screen), that music source will play instead of the music source associated with the active VDN.

  The COR assigned to the VDN must permit music-on-hold.

- CMS

  You can administer CMS to provide CMS reports to each tenant. You can restrict each CMS login to control, on a permission basis, only those entities that are assigned to a particular tenant. Outputs to separate printers allow any tenant to print their own CMS reports. The tenant-partitioning provider must administer CMS to provide this separation of tenant permissions.

- Dial Access to Attendant

  When a tenant dials an attendant, it accesses its own assigned attendant group.

- Emergency Access to the Attendant

  When a tenant dials emergency access, it accesses its own assigned attendant group.

- Expert Agent Selection (EAS)

  For agents in an EAS system, the Class of Restriction assigned to the logical agent ID (not the physical extension) determines whether callers on hold can hear music.

- Hunt groups

  The tenant number assigned to the hunt group extension determines the music source callers to the hunt group hear while they're in queue or on hold.

- Intercept Treatment

  When access to the attendant is designated as intercept treatment, the caller accesses their assigned attendant group.

- Malicious Call Trace

  By default, Malicious Call Trace extensions are assigned to tenant partition 1. Therefore, if Malicious Call Trace is enabled, any telephone with permission to call tenant partition 1 can use it.

- Multiple Listed Directory Numbers

  Each Listed Directory Number is assigned to a tenant partition.

- Multiple Audio/Music Sources for Vector Delay

  When an audio source is specified by a *wait-time <time> hearing <treatment>* vector step, the audio source assigned to the tenant number of the active VDN is the one that plays.

  If you use a *wait-time <time> hearing <extension> then <treatment 2>* command where the *<extension>* is an audio source (assigned on the Announcements/Audios Sources screen), that audio source will play instead of the one associated with the active VDN. For information on administering multiple audio sources see the *Avaya MultiVantage™ Call Center Software Release 11 Call Vectoring and Expert Agent Selection (EAS) Guide*.

- Music-on-Hold Access

  When Tenant Partitioning is enabled, you can assign a unique source for music to each tenant. If Tenant Partitioning on the System-Parameters Customer-Options screen is set to **y**, you must use the Music Sources screen to administer music-on-hold.

- Night Service

  Each tenant can have its own Listed Directory Number (LDN) night destination, Trunk Answer on Any Station (TAAS) port, or night attendant.

- PC Interfaces

  You must assign each PC interface to a tenant partition.

- PC/PBX Connections

  You must assign each PC/PBX Connection to a tenant partition.

- PC/ISDN

  You must assign each PC/ISDN to a tenant partition.

- Remote Access

  You must assign each Remote Access barrier code to a tenant.

- Traffic Studies

  Traffic studies do not report the tenant partition number of the extension or trunk group used. You must infer the tenant partition number must be inferred from the extension or trunk-group number.

- Uniform Dial Plan

  If a Uniform Dial Plan is in place between switches, tenant partition identification is not passed between the switches, and so tenant-partition restrictions are not enforced between the switches without special administration.

  NOTE:
  Tenant Partitioning restrictions do not override COR restrictions. COR restrictions are independent of tenant partitions.

Terminal Translation Initialization

Terminal Translation Initialization (TTI) allows you to merge an x-ported extension to a valid port by dialing a system-wide TTI security code and the extension from a telephone connected to that port. TTI also allows you to separate an extension from its port by dialing a similar separate digit sequence. This action causes the extension to be administered as an X port.

When TTI is enabled for voice, all voice ports (except Basic Rate Interface (BRI) ports) become TTI ports or ports from which a TTI merge sequence can occur.

TTI is usually used to move phones, however, it also supports connecting and moving attendants, data modules, voice/data telephones, and ISDN-BRI telephones.

Attendants

In order for attendants to use TTI, you must assign an extension to the attendant console. TTI port translations are the same for digital telephones and attendant consoles. To merge a digital TTI voice port and an attendant, you must first administer the attendant as an X port. Then a digital telephone must be plugged into the jack assigned to the attendant console, and the TTI merge digit sequence must be entered on the digital telephone. Once the TTI merge has been completed for the attendant console, the digital telephone must be unplugged and the attendant plugged into the jack.

An attendant console can be separated from its port only through administration. A TTI separate request from an attendant console gives the user intercept treatment.

Data modules

Data modules have different tones and digit sequences.

In the merge and separate control flow, different tones are given to the telephone user to give the current status of the TTI operation. Instead of audible tones, status messages are displayed on a telephone connected to a data module when activating the TTI sequence through keyboard dialing. If the TTI State field is set to **data**, you see the data display messages. If the TTI State field is set to **voice**, you hear the tones.

For a stand alone data module, the TTI merge/separate digit sequence is entered in one line at a dial prompt:

- DIAL: <TTI feature access code><TTI security code><ext>

Separate prompts are not given for the TTI security code and extension.

Voice/data telephones

A telephone with a data terminal (DTDM) is treated as a telephone in the TTI merge and separation sequence. The DTDM is merged with and separated from its hardware translation at the same time the telephone is merged or separated. The TTI merge and separate sequence can only be initiated through the telephone for DTDMs; it cannot be initiated through the data port.

ISDN BRI telephones

The TTI separation sequence for Automatic-TEI SPID-initializing BRI telephones is identical to the sequence used for telephones. However, the merge sequence is different.

- Separation sequence
  1. Feature Access Code
  2. Security Code
  3. Extension
- Merge sequence
  1. Connect the telephone to any port to get power.
  2. Program the SPID to the extension with which it will be merged.
  3. Unplug the telephone (this is necessary even if the telephone is connected to its intended port).
  4. Connect the telephone to its intended port (this port should indicate Equipment Type: TTI Port).
  5. Receive dial tone.
     — If there is dial tone, the merge is complete.
     — If there is no dial tone, the telephone's SPID is not an available extension.

You can dial the TTI merge sequence for BRI sets only if a user separates a BRI extension from its set and then wants to undo the process by reassociating the set to the same extension. Note that you cannot use the SAT to put an X in the port field of a BRI set that is still connected to the switch. You must use the TTI separation sequence from the set.

Analog Queue Warning Ports and External Alert Ports

The analog queue warning port (used for hunt groups) and the external alert port may be administered with an X in the Port field. These extensions can be merged to an analog port via TTI. The merge must be done by an analog set, and then the analog set is unplugged from the port. These extensions cannot be separated from their port location with the TTI feature. A TTI separate request from one of these ports gives you intercept treatment.

Security measures

SECURITY ALERT:

If you do not manage this feature carefully, its unauthorized use may cause you security problems. For example, someone who knows the TTI security code could disrupt normal business functions by separating telephones or data terminals. You can help protect against this action by frequently changing the TTI security code. You can further enhance system security by removing the Feature Access Code (FAC) from the system when it does not need to be used (for example, there are no moves going on at present). Consult the Avaya Products Security Handbook for additional steps to secure your system and find out about obtaining information regularly about security developments.

Interactions

- Attendant

  It is possible to have the attendant in Position Available Mode and still separate. Any calls queued, held, or seen as active for the attendant prevent separation.

- Attendant Night Service

  The night service station cannot be separated while in night service.

- Attendant Release Loop Operation

  If the attendant separates before the attendant-timed reminder-interval expires, all calls held with the release loop operation by the attendant are reclassified as attendant group calls.

- Automatic Callback

  If a telephone has Automatic Callback active for another telephone, executing TTI separate for either telephone breaks the automatic callback sequence.

- Call Coverage

  If a telephone separates while Send All Calls or Goto Coverage is active, these features remain active while the telephone has no associated hardware.

  You can separate a telephone that is the target of Send All Calls or Goto Coverage; the features function as if the telephone were busy.

- Call Coverage Answer Group

  If a extension was an X port, then rejoins the group as a result of a TTI merge, a PSA associate, or a port assignment, that telephone is excluded from all transactions already active in the call coverage answer group.

- Call Forwarding

  A telephone can separate while Call Forwarding is active. If a destination extension for call forwarding separates, Call Forwarding to that extension remains active. Calls forwarded while the telephone is separated hear a busy signal.

- Call Pickup

  If a line appearance is available, a member of a call pickup group may separate at any time. If a call is attempting to terminate, and a member of a group associates, that member does not join the group for the call that is currently in progress, but is available for all subsequent calls to that group.

- Expert Agent Selection

  Station user records cannot be shared between TTI ports and EAS login ID extensions. This reduces the number of possible TTI ports your system provides, depending on the number of administered EAS login IDs. For example, if you administer 2,000 EAS login IDs, the maximum number of TTI ports that the system can provide is reduced by 2,000.

- Hunt Group Uniform Call Distribution/Direct Department Calling

  The system excludes telephones previously X-ported as a result of a TTI separate, a PSA dissociate, or administration from all transactions already active in the hunt group when the telephone is merged.

- Site Data

  If Terminal Translation Initialization is enabled, a warning message displays when you tab off the Site Data fields.

  If Terminal Translation Initialization is enabled, and you change the Port field on the Station screen from “x” to a real port number and change the Room, Jack, or Cable fields in the Site Data section, a warning message displays when you tab off the fields.

- Terminating Extension Group

  If any member of the TEG (that was previously an X port as a result of TTI, PSA or telephone administration) is merged, that member is excluded from all transactions already taking place in the TEG when that member is merged. The member is able to join in all subsequent calls to the group.

Related topics

For more information about configuring the TTI system-wide fields, refer to ''Feature-Related System Parameters'' on page 766.

For more information about setting the TTI feature access codes, refer to ''Feature Access Code'' on page 752.

Terminating Extension Group

A Terminating Extension Group (TEG) allows an incoming call to ring as many as 4 phones at one time. Any user in the group can answer the call.

Detailed description

You can administer any phone as a TEG member; however, only a multiappearance phone can be assigned a TEG button with merged-status lamp. The TEG button allows the user to select a TEG call appearance for answering or bridging onto an existing call but not for call origination.

When a TEG members answers an incoming call, a temporary bridged appearance is maintained at the multiappearance phones in the group. However, this appearance is not visible. Any TEG members can bridge onto the call by pressing the TEG button.

Considerations

- A phone user can be a member of more than one TEG, but can have only one TEG button for each group.
- A TEG can handle only one TEG call at a time. Additional calls do not reach the TEG. If a coverage path is assigned to the TEG, the additional calls route accordingly.

Interactions

- Automatic Callback

  This feature cannot be active for a TEG.
- Bridged Call Appearance

  Calls to a TEG cannot be bridged, except via a Temporary Bridged Appearance.

- Call Coverage

  A TEG can have a Call Coverage path assigned, but cannot be a point in a Call Coverage path.

  A Send Term button for the TEG can be assigned to group members who have multiappearance phones. When a user presses Send Term, calls to the TEG redirect to coverage. The merged status lamp lights on all phones with a Send Term button. Any member with a Send Term button can deactivate Send Term by pressing the button. Incoming calls are directed to the group.

- Call Park

  A TEG call cannot be parked on the group extension. However, a group member answering a call can park a TEG call on their own extension.

- Direct Department Calling and Uniform Call Distribution

  A TEG cannot be a member of a DDC or UCD group.

- Internal Automatic Answer

  TEG calls are not eligible for Internal Automatic Answer; however, calls placed to an individual extension are eligible.

- Leave Word Calling

  Leave Word Calling messages can be stored for a TEG and can be retrieved by a member of the group, a covering user of the group, or a system-wide message retriever. Phone Display and proper authorization can be assigned to the message retriever. Also, a remote Automatic Message Waiting lamp can be assigned to a group member to provide a visual indication that a message has been stored for the group. One indicator is allowed per TEG.

- Privacy — Manual Exclusion

  Privacy — Manual Exclusion can be assigned to any of the phones in a TEG to prohibit bridging by other group members. A TEG member who attempts to bridge onto a call with Privacy — Manual Exclusion active is dropped.

- Temporary Bridged Appearance

  At multiappearance phones in the TEG, a temporary bridged appearance is maintained after a call is answered. Thus, other members of the group can bridge onto the call.

Related topics

Refer to ''Terminating Extension Group'' on page 1156 for information about and fields descriptions on the Terminating Extension Group screen.

Refer to ''Station'' on page 1066 for information on button assignments.

Refer to ''Assigning a terminating extension group'' on page 240 for instructions.

Time of Day Routing

You can use Time of Day Routing to select route patterns for calls according to the time of day and day of the week. You need to define the route pattern you want to use before you set up time of day routing.

You can route calls based on the least expensive route, and you can deny outgoing long-distance calls after business hours to help prevent toll fraud. You can use partition groups to assign different time of day route plans for different groups of users.

Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS) must be administered on your switch before you use Time of Day Routing. Time of Day Routing applies to all AAR or ARS outgoing calls and trunks used for call forwarding to external numbers.

Interactions

- DCS

  Be careful when you make Time of Day Routing assignments in a DCS environment. The user's COR Time of Day Plan Number determines whether or not the call is routed to a DCS trunk group. The call loses feature transparency if it is not routed to a DCS trunk group.

  When a call routes over a DCS trunk, the switch at the far end routes the call according to the COR Time of Day Plan Number of the incoming trunk.

- Remote Access

  When remote access is used and an authorization code or barrier code is dialed for an AAR or ARS call, the COR Time of Day Plan Number of the barrier code or authorization code is used to route the call.

Related topics

- Refer to ''Setting up time of day routing'' on page 264 for information on setting up Time of Day routing.
- Refer to ''Defining ARS Partitions'' on page 261 if you want to set up different Time of Day routing for different partition groups.
- Refer to ''Automatic routing — general'' on page 1383 for information on AAR and ARS

Time of Day Clock Synchronization

Time of Day Clock Synchronization provides you with the capability of synchronizing your MultiVantage system clock(s) with Internet servers providing Coordinated Universal Time (UTC).

Time of Day Clock Synchronization enables a switch to synchronize its internal clock to UTC time provided by Internet time servers. The LINUX or Windows 2000 platforms, running NTP or SNTP software, poll the time servers for the UTC time. UTC time is then converted to the local time of the switch. The platform system clock then provides the synchronized time to MultiVantage.

Detailed description

UTC time is received from time servers connected to the Internet. LINUX/Windows 2000 connect, through an Ethernet port, to these time servers using NTP or SNTP software. UTC time then is converted to the local time of the switch.

ORYX/PECOS and Legacy DEFINITY

The existing design of legacy DEFINITY uses a software clock to keep track of the time of day. A MultiVantage SAT command is used to set and display this time. A hardware clock also is provided to maintain the system when power is lost.

The software and hardware clocks are set whenever a manual change is made on the Set Time SAT screen or when local time changes because of daylight savings time adjustment. These adjustments can have significant delays (more than a minute) in a heavily loaded system because of process priorities so the accuracy of the time actually can be compromised by a manual set of the time.

The software clock is not synchronized with an external timing source so this time will draft over time. Corrections for the drift of the software and hardware clocks can be made by software on LINUX but not on Oryx/Pecos.

S8100 Media Server and Windows 2000

The system time is maintained by the operating system at UTC. Library functions are provided to obtain local time. The time zone and use of daylight savings time are set from the time/date dialog box on the desktop.

SNTP compliant software is available on Windows 2000. This software runs a system service that can be started up at system boot time. Windows 2000 uses Registry entries to set SNTP parameters. Avaya will not support the optional use of a Microsoft network domain controller as the primary time server.

Use of Time of Day information by MultiVantage

MultiVantage features use the time of day information for a variety of reasons.

- Time stamp data elements (Call Detail Recording records, error logs, Message Sequence Trace data, Malicious Call Trace records, BCMS, and Call Management System data)
- Set time-out intervals (Automatic wakeup messages, Do no disturb intervals)
- Schedule tasks (scheduled maintenance, moving CDR data to the output file)
- Synchronize time of day information with other processors (periodic TOD messages sent to the 860 processor on the S8100 Media Server processor circuit pack)

The time of day value used for the MultiVantage system time is essentially local time with or without daylight savings time adjustments.

S8100 Media Server uses the Windows 2000 and Avaya S8700 Multi-Connected Media Server uses LINUX as the platform operating system. The use of the system clock from the platform OS is transparent. Both Windows and LINUX use UTC for the system (software) clock. Library functions and APIs are used to express the time in local time. MultiVantage code still uses local time. However, the daylight savings rule for system time is controlled by the OS. This is accounted for by using the time adjustments from the OS for Daylight Savings Rule 1. The display of this rule on the System Access Terminal Daylight Savings Rules screen is display-only. The values displayed for the time adjustments are read from the operating system.

In legacy DEFINITY when a change is made to the system time because of a manual change on the Set Time SAT screen or because of an adjustment for daylight savings time, certain DEFINITY applications that need to know about the change are notified before the change is made to the system clock.

If you do not have Internet access, you can continue to manually set the time of day.

CAUTION:

Avoid setting the time manually while the TOD clock sync feature is enabled. Manually setting the time, while NTP/SNTP is running, defeats the purpose of synching the UTC time and could cause problems depending on the difference between the system time and UTC.

Setting time of day clock synchronization

The recommended configuration for ASA PC and time synchronization:

1. The PC clock is to be set to the local time including applying the local daylight savings time (DST) rule.
2. The offset for each switch is to be set based on the difference between the local ASA PC standard time and the switch standard time (for example, use offset of "plus 1 hour" if ASA PC standard time is 02:00 when the switch standard time is 03:00 per existing operation.
3. Schedule the task to occur no more often than every 24 hours at a low traffic time period based on the local time at the switch and in the middle of a BCMS/CMS interval being used at the switch (for example, 11:07 PM or 3:07 AM). Also the time picked should not be set to a time period when MultiVantage nightly maintenance is scheduled to be run or the BCMS next day period starts. Note that the time ASA uses to decide when to send the time change is based on the local PC time including DST when applicable. Therefore the start time and interval entered for a switch must be based on the switch local time converted to the PC local time.

   Running the task more frequently (for example, every 12 hours) makes it more difficult to have it run during a low traffic period or it could cause the scheduled time to occur at different times during the day as time goes on (for example, picking any entry other than Every 24 hours or 12 hours will make the scheduled time "creep" over the day by the selected time interval since only the start date-time and repeat interval can be specified). The use of the Weekly or Monthly Recurrence Patterns does not have the "creep" problem.
4. Note that ASA must be running and OSSI connectivity to the switches must be available (via LAN or dialup) during the scheduled times for synchronizing the MultiVantage switches that it manages.

The recommended configuration for each MultiVantage switch time synchronization:

1. The system clock may be set to either local time using automatic DST changes, local standard time (without DST changes - DST Rule set to 0) or GMT time (without DST changes). In the case where automatic DST changes is not assigned to the system clock (that is, the DST rule is set to 0 on the Date and Time screen), you can use the Multiple Locations feature to display DST (and local time in the case where the system time is set to GMT) on the station sets homed on the local port networks. However, all switch reports (including BCMS), CMS/Supervisor reports, TOD routing and scheduled events will be based on the system clock time not the location time. It is expected that in most cases the system clock will be set to local time with DST change. In each case, the ASA assigned offset for the switch will provide the proper conversion from PC time to the desired system clock standard time.

2. In all cases the Multiple Locations feature provides local time display on station sets based on the local time zone and DST rule in effect at the remote port network (EPN) site. The time determined by the location assignments also applies to do-not-disturb, Automatic Wakeup programming and execution (however, the announced time is the system time). Use the Locations screen to define the time zone offset from the standard time local to the switch location (the PPN location) and the DST rule to be used for any remote EPNs that are located in different time zones. The location numbers established in the Locations screen then are assigned to the appropriate EPN cabinets using the cabinet description screen. The same location number can be assigned to more than one cabinet that are in the same time zone. Note that only R (Mips) configurations can be assigned locations other than 1 (the csi/si configuration defaults to location 1).

3. Each of the configuration possibilities is summarized as follows:

   a. **System time set to local time with automatic DST changes** (when observed in that area) - Use the Date and Time screen to set the time for the local time zone and the DST rule defined on the ‘‘Daylight Savings Rules’’ screen. The time initially can be set manually to the local standard time during fall/winter/spring or to the local DST time in the summer. If DST is not observed then the following configuration b would be used. All reports and display station sets will show the local time with the DST rule applied. TOD routing/scheduled events occur based on the system clock (which will be in the local standard time or local DST depending on the time of the year). Use the Multiple Locations feature to display local time on station sets at remote sites.

b. **System time set to local time without DST rules -** the Date and Time screen has the standard time for the local time zone set and the DST rule field is set to **0**. All reports and TOD routing/scheduled events are all based on local standard time regardless of whether DST is being observed or not. Display station sets at the local PPN and remote EPN sites can have actual local time (with DST applied) via the Multiple Locations feature.

c. **System time set to GMT (UTC) -** the Date and Time screen has GMT standard time set with the DST rule set to 0. All reports and TOD routing/scheduled events are all based on GMT time. Display station sets at the local PPN and remote EPN sites can have actual local time (with DST applied) via the Multiple Locations feature. The ASA offset for this MultiVantage switch will be the offset of the ASA PC local standard time from GMT/UTC time (for example if the ASA PC is in Denver, the offset for any switch using GMT time is +7).

NOTE:
When NTP starts up, there will be a gap of possibly several minutes between MultiVantage time and the time provided by NTP. NTP reduces this gap to milliseconds over a period of hours.

Transfer

Transfer allows telephone users to transfer trunk or internal calls to other telephones or trunks without attendant assistance.

Considerations

- You can administer transferred-trunk calls to receive either music or silence if the first part of the transfer places the call on hold.
- Multi-appearance telephones must have an idle appearance to transfer a call.
- Single-line telephone users momentarily flash the switchhook or press the RECALL button, dial the desired extension, and hang up. Multi-appearance telephone users press the TRANSFER button, dial the desired extension, and press the TRANSFER button again.
- If, on the Feature-Related System Parameters screen, the Transfer Upon Hang-up field is **y**, users can transfer a call by pressing the TRANSFER button, dialing the desired extension, and then hanging up. The user can hang up while the desired extension is ringing or after the party has picked up. The user also can still press the TRANSFER button a second time to complete the transfer process.

- If, on the Feature-Related System Parameters screen, the Abort Transfer field is **y**, users can abort the transfer a call by pressing the TRANSFER button, dialing the desired extension, and then hanging up or selecting any non-idle call appearance. The user must press the Transfer button again to complete the process (see Note). If the user selects an idle call appearance, the transfer still is active.

  NOTE:
  If both the Abort Transfer and Transfer Upon Hang-Up fields are **y** and you press the TRANSFER button and then dial the complete transfer-to number, hanging up the phone transfers the call.

- Users of DCP, Hybrid, and wireless phones can transfer a call on hold without removing the call from hold. If there is only one call on hold, no active call appearances, and an available call appearance for the transfer, the user can transfer the call simply by pressing the Transfer button. Avaya MultiVantage assumes the transfer is for the call on hold, and the transfer feature works as usual.

  If there is more than one call on hold, the user must make a call active in order to transfer it. If the user presses the Transfer button with two or more calls on hold, Avaya MultiVantage will ignore the transfer attempt since it will not know which call the user wants to transfer. If there are calls on hold and an active call, pressing the Transfer button will start the transfer process for the active call.

- Avaya MultiVantage can be administered to display a confirmation message to users upon successful call transfers. The confirmation message will only be visible to users with DCP, Hybrid, wireless (except for 9601), or ISDN-BRI display phones. All of these phones, except for the Hybrids, can display the confirmation message in English, Spanish, French, Italian, or a language you define. Hybrid phones only display the message in English.

- You can administer the system to return a transferred call to the originator if the transferred-to party does not answer within a set time limit. To do this, enter a value in the Station Call Transfer Recall Timer field on the Feature-Related System Parameters screen.

Interactions

- Attendant Conference

  This may not operate properly if the CO does not provide answer supervision. In that case, the Answer Supervision Timeout and Outgoing End of Dial fields (on the CO Trunk Group screen) must be set to the same non-zero number. The Receive Answer Supervision field must be set to **n**.

  If the CO does provide answer supervision, the Answer Supervision Timeout and Outgoing End of Dial field must be set to **0** and the Receive Answer Supervision field must be set to **y**.

- Bridged Call Appearance

  A bridged call appearance can be used to transfer calls.

- CentreVu Agent

  CentreVu Agent supports the conference and transfer of calls on hold, but it does not display the confirmation message for transferred calls.

- Class of Restriction

  Set the Block Transfer Display field on the COR screen to **y** if you do *not* want users to receive a confirmation message for call transfers.

- Integrated Services Digital Network Basic Rate Interface (ISDN-BRI)

  When an ISDN-BRI telephone, assigned with the Select Last Used Appearance field set to **y**, completes a transfer while off-hook using the handset, the user is left listening to dial tone on the last-used appearance.

- Internal Automatic Answer

  Transferred calls can be answered automatically via IAA.

- QSIG Networking

  If calls over an ISDN-PRI trunk are administered for Supplementary QSIG, then additional call information may display.

- Station

  When a multifunction telephone dials enough digits to route a call, but the call could route differently if additional digits were dialed, the telephone does not recognize the Conference or Transfer buttons. The user must delay dialing for 3 seconds or dial # to indicate that the call can be routed based on the digits already dialed. The Conference or Transfer buttons then are recognized and the switch completes the operation.

Pull Transfer

Pull Transfer allows either the transferring or transferred-to party to press the Transfer button to complete the transfer operation.

When attendants control calls, called parties cannot use Pull Transfer. Attendants who are called parties cannot use Pull Transfer. When attendants have parties on hold they are transferred with the standard transfer process.

To use Pull Transfer, calling parties and called parties must be on the same switch, or called parties must be reached via Italian TGU/TGE tie trunks.

Called parties using analog telephones flash the switchhook or press the flash key or recall button to transfer calls. Called parties using digital phones press the transfer key to complete transfers.

Interactions

- Analog Station Recall Operation and Feature Activation

  When called parties initiate either analog-telephone recall or feature activation, callers are not put on hold for transfer, they are transferred via Pull Transfer.

- BRI telephones

  Callers using BRI Stations reach desired parties through an intermediate step by calling a party who calls a final destination. Intermediate parties activate pull transfer to complete transfers. Final called parties go off hook as if a new transfer was originated.

- Call Detail Recording

  The switch checks to ensure that calls are correctly recorded with CDR when Pull Transfer is completed.

- Digital Station Transfer Operation

  When called parties initiate transfer operations, callers are not put on hold for transfer; they are transferred via Pull Transfer.

- Non-BRI telephones:

  Callers using Non-BRI telephones reach desired parties through an intermediate step by calling a party who calls a final destination. Each called party activates pull transfer.

Related topics

- "Feature-Related System Parameters" on page 766
  - Music on Transferred Trunk Calls field
  - Intercept Treatment on Failed Trunk Transfers field
  - Abort Transfer
  - Transfer Upon Hang-Up
- "Class of Restriction" on page 636
  - Block Transfer Display field

Transfer — Outgoing Trunk to Outgoing Trunk

Outgoing Trunk to Outgoing Trunk Transfer (OTTOTT) permits a controlling party (such as a station user or attendant) to initiate two or more outgoing trunk calls and then connect the trunks. This operation removes the controlling party from the connection and conferences the outgoing trunks. Alternatively, the controlling party can establish a conference call with the outgoing trunks and then drop out of the conference, leaving only the outgoing trunks on the conference.

NOTE:
This is an optional enhancement to Trunk-to-Trunk Transfer and requires careful administration and use. Distributed Communication System (DCS) Trunk Turnaround may be an acceptable and safer alternative to this feature.

OTTOTT allows calls to be established in which the only parties involved are external to the switch and are on outgoing trunks. This type of call can result in locked-up trunks, such as trunks that cannot be disconnected except by busying-out and releasing the affected trunk circuit. To clear the lockup, a service technician must reseat the trunk board, or busy-out and release the affected trunk.

Detailed description

This feature is enabled by administering the Disconnect Supervision Out field to **y** on at least one outgoing trunk group screen, and enabling Trunk-to-trunk transfer on the Feature Related System Parameters screen. In addition, the network must provide answer supervision. The answer supervision test increases the probability, but does not guarantee, that a disconnect signal is received from the remote end of the trunk. To mitigate problems associated with its accidental use, this feature is administrable only on trunk groups on the Trunk Group screen. It is not a system option.

DCS networks provide a similar but more restrictive version of this feature, called DCS Trunk Turnaround, which permits two outgoing trunks to be connected when the switch at the remote end of one of the trunks agrees to turn around the logical direction of the trunk. DCS trunk turnaround is permitted, when some other party involved in the call (at the remote switch) can provide disconnect supervision.

Without OTTOTT or DCS, a conference involving two or more outgoing trunks is permitted only when at least one remaining conference party is an attendant, incoming trunk, or station.

Considerations

- OTTOTT is not intended for use in DCS networks, since DCS Trunk Turnaround provides comparable capabilities much more safely. However, use of OTTOTT with DCS is not prohibited, and may be useful when one or more of the trunks goes off the DCS network.

Security Measures

SECURITY ALERT:

This feature can be used to transfer an outside party to a trunk over which toll calls might be made, and enabling it poses significant security risks. Since trunks have to be specifically administered for OTTOTT, you should examine the Class of Restriction (COR) and Facility Restriction Level (FRL) of that trunk group to determine if they are appropriate.

SECURITY ALERT:

OTTOTT is not a system-wide parameter. It is administered on a trunk-group basis. You must enable the Trunk-to-Trunk Transfer field on the Feature-Related System Parameters screen for this feature to work. If you deem that the feature is not relevant to your business practices, do not enable it. Alternately, if a temporary need presents itself, you can temporarily enable this feature and then turn it off.

Interactions

- DCS Trunk Turnaround

  OTTOTT increases the set of cases in which DCS Trunk Turnaround may be accepted. However, use of OTTOTT in combination with a DCS network is strongly discouraged. The following algorithm describes the DCS Trunk Turnaround request process.

  a. If any party on the call receives a local-dial, busy, intercept, or reorder tone, deny turnaround.

  b. If any remaining party is an answered station or attendant, accept turnaround.

  c. If any remaining party is on an incoming trunk, accept turnaround. For the purposes of this check, an outgoing DCS trunk that has been turned around an odd number of times via a DCS trunk turnaround is considered an incoming trunk with disconnect supervision. Similarly, an incoming DCS trunk that has been turned around an odd number of times is considered an outgoing trunk.

  d. If any remaining party is an outgoing trunk administered for OTTOTT that has received answer supervision, accept turnaround.

  e. If any remaining party is an outgoing DCS trunk, forward the turnaround request.

  f. Otherwise, deny turnaround.

- Incoming Disconnect Supervision

  Outside of the U.S., incoming disconnect supervision is a switch capability that restricts transfers or conferences for certain incoming trunks. In the U.S., all incoming trunks are assumed to provide disconnect supervision. In some countries this assumption is not valid, so administer whether or not an incoming trunk provides disconnect supervision for each trunk group.

- Personal Central Office Lines

  Transfer of personal CO lines (PCOLs) is not subject to the normal restrictions applied to transfer of other trunks. These transfers are allowed since the PCOL appearance remains on one or more stations as a feature button. System users must be aware that the DROP button cannot be used to disconnect the transferred-to party from the call. Hence, if an outgoing PCOL is transferred to an outgoing trunk and neither of the trunks can supply a disconnect signal, the two trunks lock up.

- QSIG Global Networking

  If either call is over an ISDN-PRI trunk administered with Supplementary Service Protocol b (QSIG), additional call information may display.

- Release Link Trunks

  RLTs are used by Centralized Attendant Service (CAS). An outgoing RLT at a remote branch is used to access an attendant at the main. The attendant at the main can transfer the incoming caller to a station or trunk at the branch. The RLT is typically used only for a short period of time and is usually idled after the transfer is established.

  A station at a branch can transfer an outgoing trunk to the attendant at the main. This transfer could be viewed as an OTTOTT (the attendant is accessed via an outgoing RLT). Since administering outgoing disconnect supervision for RLT trunks provides no additional capability, this administration is not provided for RLT trunks.

- Restriction

  Restrictions on the transferring party may block a transfer or drop operation even when Outgoing Disconnect Supervision is provided.

- Trunk-to-Trunk Transfer

  If this feature-related system parameter is set to restricted, all trunk-to-trunk transfer/release/drop operations for public trunks (CO, CPE, CAS, DID, DIOD, FX, and WATS) have calls terminated or receive denial. If the parameter is set to none, all trunk-to-trunk transfers (except CAS and DCS) have calls terminated or receive denial.

  Hence, this option must be set to all to enable OTTOTT operation for these types of trunks. The number of public-network trunks allowed on a conference call is administrable. This number defaults to 1, so if OTTOTT is being used to connect two or more public network trunks, you must increase this limit on the Feature-Related System Parameters screen.

- Trunks (CO, FX, and WATS)

  You cannot have two CO, FX, or WATS trunks in a OTTOTT connection, even if the Disconnect Supervision - Out field is set to **y**.

Transfer — Trunk-to-Trunk

Trunk-to-Trunk Transfer allows the attendant or user to connect an incoming trunk call to an outgoing trunk.

SECURITY ALERT:

Trunk-to-trunk transfer poses a significant security risk. Use this feature only with extreme caution.

The system provides three levels of administration for this feature: system-wide, COR-to-COR, and COS.

To administer Trunk-to-Trunk Transfer system-wide, complete the Feature-Related System Parameters screen. To restrict Trunk-to-Trunk Transfer on a trunk-group basis, assign COR-to-COR calling-party restrictions on the Class of Restriction (COR) screen. To allow individual users to control Trunk-to-Trunk Transfers, assign capabilities on the Class of Service (COS) screen.

Considerations

- Trunk-to-Trunk Transfer is particularly useful when a caller outside the system calls a user or attendant and requests a transfer to another outside number. For example, a worker, away on business, can call in and have the call transferred elsewhere.
- Transferred trunk calls can be administered to receive either music or silence.
- Some central office (CO) trunks do not signal the PBX when the CO user disconnects from a call. The system ensures that incoming CO trunks without Disconnect Supervision are not transferred to outgoing trunks or to other incoming CO trunks without Disconnect Supervision.
- An attendant-assisted call connecting an outgoing trunk or incoming trunk without Disconnect Supervision to an outgoing trunk must be held on the console. The system does not allow the attendant to release the call. The attendant can, however, use the Forced Release button and disconnect all parties associated with the call.
- If a user has connected two outgoing trunks or an outgoing call and an incoming call without Disconnect Supervision, the user must remain on the call. Otherwise, the call is dropped. An incoming trunk with Disconnect Supervision can be connected to an outgoing trunk without the user remaining on the call. An incoming trunk can also be connected to another incoming trunk without the user remaining on the call if one of the incoming trunks has Disconnect Supervision.

Interactions

- Attendant Lockout

  Attendant Lockout does not function on Trunk-to-Trunk Transfer.
- Call Vectoring

  Station control of Trunk-to-Trunk Transfer does not affect routing of incoming trunks to a VDN that ultimately routes to a destination off-net.

  A route to a number off the switch does *not* require you to enable trunk-to-trunk transfer.

- Tenant Partitioning

  Station control of Trunk-to-Trunk Transfer is prohibited between trunks in different tenant partitions if those partitions are restricted.

Trunk Flash

Trunk Flash allows a feature or function button on a multifunction telephone or attendant console to be assigned as a Flash button. Pressing this button while connected to a trunk (which must have been administered to allow trunk flash) causes the System to send a flash signal out over the connected trunk.

Trunk Flash enables multifunction telephones to access central office customized services that are provided by the far-end or Central Office (CO) located directly behind the MultiVantage System. These central office customized services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from the MultiVantage system telephone on an active trunk call. The Trunk Flash feature can help to reduce the number of trunk lines connected to the MultiVantage switch by:

- Performing trunk-to-trunk call transfers at the far-end or CO, which eliminates the use of a second trunk line for the duration of the call and frees the original trunk line for the duration of the call.
- Performing a conference call with a second outside call party, which eliminates the need for a second trunk line for the duration of the call.

> **NOTE:**
> Some analog Dual-Tone Multi-Frequency (DTMF) telephone sets used in Italy and the United Kingdom are equipped with a FLASH button that, when pressed, generates a rotary digit 1. When an analog telephone which is administered as a DTMF telephone (for example, as a 2500 or 71nn-type telephone) transmits a rotary digit 1, the system treats the signal as a recall signal from the telephone set to the Avaya MultiVantage.

When used by a Centralized Attendant Service (CAS) attendant connected to an Release Link Trunk (RLT), the flash controls certain CAS features at the branch. When used by a multifunction telephone or non-CAS attendant connected to a CO, Foreign Exchange (FX), or Wide Area Telecommutings Service (WATS) trunk, the flash controls certain features (such as add-on) at the connected CO.

Trunk Flash is not available on Personal Central Office Line (PCOL) groups.

The system supports the Trunk Flash signal for incoming, outgoing, or 2-way call directions on selected 2-wire analog (ground-start or loop-start) or digital (DS1) trunks or Tie trunks on DS1.

If the trunk group is a DS1 trunk in Italy, the Trunk Flash feature applies only to outgoing calls.

If the trunk is not directly connected to the far end or CO providing the customized services, use of the Trunk Flash signal may cause the call to be disconnected by the far end or CO.

Calls made after the Flash are not recorded in Call Detail Recording (CDR) records.

Considerations

NOTE:
The Trunk Flash feature allows the telephone user to receive central office dial tone, and place a call that is not monitored by the MultiVantage system, and is not subject to restrictions (toll, FRL, COR, etc.). Therefore you should enable this feature with caution.

- A Trunk Flash button can be assigned on CAS attendant consoles, non-CAS attendant consoles, and multifunction telephones. For CAS attendants, use of this button is limited to certain CAS features via RLT trunks. For multifunction and non-CAS attendant consoles, this button is used for the Trunk Flash feature.
- FAC activation of the trunk flash feature is allowed.
- The Flash button is used by the Trunk Flash and CAS features.
- System features (such as internal conference call, transfer, and call park) may be combined with custom services (that is, CO-based features that are activated/controlled by sending a flash signal over the trunk to the CO). However, mixing Avaya MultiVantage features with custom services causes complications for the user when tracking a call. MultiVantage Systems cannot give the local telephone user status information on the custom services.
- The Trunk Flash feature may only be accessed if the call has only one trunk, the trunk must be outgoing from the PBX's perspective, and the trunk group of that trunk has Trunk Flash enabled. The Trunk Flash feature is disabled when the call involves more than one trunk, even if all the trunks have Trunk Flash enabled.
- Any Avaya MultiVantage telephone can flash and dial a FAC to access the Trunk Flash feature; any Avaya MultiVantage telephone with a flash button can access the Trunk Flash feature by hitting the flash button. The system allows as many as five telephones to participate in a conference call with the trunk line party. However, to access the Trunk Flash feature, at least one of the telephones must have a Flash button.

- In a call involving more than one telephone, one of the telephones may press the Flash button, and another telephone may dial the phone number. The telephone that dials the phone number is not required to have a Flash button.
- If the far-end/CO does not support custom services, the call may be dropped by the far-end/CO on sending the flash signal or the signal may be ignored and a click-click sound is heard.

Trunks and Trunk Groups

Trunks connect Avaya MultiVantage to other pieces of equipment (adjuncts) and to other switches. In general, trunks connect your switch to one of four things:

- the public telephone network
- a private telephone network
- the Internet or a private intranet
- switch adjuncts, such as a loudspeaker paging system or a source for music or announcements

When trunks of the same type are used for the same application, assign them to the same trunk group. A trunk group allows you to assign service characteristics to the group rather than administering each trunk individually.

NOTE:
Trunks and access endpoints consume the same resource. The sum of trunks and access endpoints cannot exceed the total number of trunks allowed on your system.

This chapter contains information about the most common analog and digital trunks. Specialized trunks such as Advanced Private-Line Termination (APLT), tandem, release-link, and DMI-BOS trunks are not covered in this manual. Refer to *Overview for Avaya MultiVantage™ Software* and *Administration for Network Connectivity for Avaya MultiVantage™ Software* for information on these trunks.

Brief description

Avaya MultiVantage supports the following basic trunk types.

Access

Used in Electronic Tandem Networks (ETN), access trunks connect satellite switches to the main switch. Unlike tandem trunks, access trunks do not carry traveling class marks (TCM) and thus allow satellite callers unrestricted access to out-dial trunks on the main switch.

CAMA — Centralized Automatic Message Accounting (E911)

CAMA trunks route emergency calls to the local community's Enhanced 911 systems and provide Caller's Emergency Service Identification (CESID) information to the system. Public Service Answering Points (PSAP) use CAMA trunks to determine the caller's physical address.

NOTE:
Avaya does not recommend tandeming 911 calls.

When the CAMA feature is administered and a 911 call is placed from an Avaya MultiVantage station, MultiVantage software searches for the proper CAMA Touch-Tone Receiver (TTR) resource in the port network (PN) where the CAMA trunk group is located. If the proper CAMA TTR resource is not available in that port network, the software checks any other available port networks for the CAMA TTR. If none is found in the entire system, a busy tone is returned to the calling party. However, if another trunk group is in the same routing pattern as the CAMA trunk group, the call routes to the next available trunk group in the pattern. For example, if a Central Office (CO) trunk group is used, then any available TTR resource can be used and the call routes as a normal call over the CO trunk and the public switched telephone network.

E911 systems use CESID information to identify the location of the calling phone. The following features can cause incorrect CESID information to be sent with a 911 call.

Bridged stations

911 calls from a bridged extension will report the CESID of the principle station.

EAS agents

911 calls from an agent in an Expert Agent Selection (EAS) skill will report the CESID of the physical station, not the logical agent.

Personal Station Access/Terminal Translation Initialization

When you use Personal Station Access (PSA) or Terminal Translation Initialization (TTI) to associate a phone with another extension, the switch will report the CESID of the extension — not the physical phone associated with it.

CO — Central Office

CO trunks transmit dialed digits for outgoing calls but not for incoming calls. Use CO trunks when you want all incoming calls go to the same destination, such as an attendant or a voice menu system. Individual users can place outgoing calls without attendant assistance. CO trunks typically connect your switch to the local central office, but they can also connect adjuncts such as external paging systems and data modules.

CPE — Customer-Provided Equipment

Use CPE trunks to connect adjuncts, such as paging systems and announcement or music sources, to the switch.

NOTE:
You can connect some types of equipment to the switch by trunks without administering a CPE trunk group. For example, trunk port interfaces can be assigned on system screens for Music-on-Hold Access (Feature-Related System Parameters screen) or voice and chime paging (Loudspeaker Paging and Code Calling Access screen).

DID — Direct Inward Dialing

DID trunks carry incoming calls directly from the local central office to your switch. These trunks transmit the digits your switch needs to route incoming calls to specific extensions. Use DID trunks when you want people calling your organization to dial individual users directly without going through an attendant or some other central point.

DIOD — Direct Inward/Outward Dialing

DIOD trunks are two-way trunks that transmit dialed digits in both directions. Thus, use DIOD trunks when you want a two-way trunk group with the convenience of direct inward dialing for incoming calls. DIOD trunks are used mostly outside North America, and typically they connect the switch to a local central office.

In North America, use tie trunks for applications that require two-way transmission of dialed digits. In the U.S., trunks that transmit digits in both directions are sometimes called “smart trunks” and are administered as T1 tie trunks.

FX — Foreign Exchange

An FX trunk is essentially a CO trunk that connects your switch directly to a central office outside your local exchange area. Use FX trunks to reduce long distance charges if your organization averages a high volume of long-distance calls to a specific area code.

IP Trunks — Internet Protocol Trunks

IP trunks allow Avaya MultiVantage to route voice calls and faxes over a local- or wide-area TCP/IP network. Use IP trunks to reduce long-distance charges by routing calls over the Internet or your intranet.

NOTE:
The origin and destination switches must both have the special hardware and software needed to route telephone calls over IP networks. See the *Administration for Network Connectivity for Avaya MultiVantage™ Software* for more information.

ISDN — Integrated Services Digital Network

ISDN trunks are digital trunks that can integrate voice, data, and video signals and provide the bandwidth needed for applications such as high-speed data transfer and video conferencing. ISDN provides end-to-end digital connectivity and uses a high-speed interface that provides service-independent access to switched services. Through internationally accepted standard interfaces, an ISDN provides circuit or packet-switched connectivity within a network and can link to other ISDN-supported interfaces to provide national and international digital communications. ISDN trunks can also efficiently combine multiple services on one trunk group.

PCOL — Personal Central Office Line

A personal central office line is one dedicated trunk that links a group of phones to the local central office or to another switch in a private network. A PCOL group can have 1–16 phones, and each phone can have one or more appearances for the personal central office line. You may administer a PCOL group as a CO, FX, or WATS trunk group. Use a PCOL line when one or more users need a direct, dedicated connection to the public network or to another switch.

Tie

Tie trunks connect a switch to a central office or to another switch in a private network. These trunks transmit dialed digits with both outgoing and incoming calls. Thus, incoming calls over a tie trunk can be routed directly to the extension the caller dialed. Tie trunks are frequently used in private networks; in addition, use tie trunks when you want a two-way trunk group with the convenience of direct inward dialing for incoming calls.

Tie trunks that connect private network switches are “universal. “This means that the trunks can be administered with a variety of translation encodes. The originating switch can recognize any start dial signal (precise dial tone, wink start, or delay dial) that the terminating switch sends. If the originating switch does not receive one of these start dial signals, it can be administered to send digits after an administered interval. (This time-out interval is the amount of time that the originating switch waits before sending digits.)

Analog tie trunks

The number of tie trunks in a connection and the technology of any multiplex systems used in the facilities affect the maximum data transfer rate. Generally, for transfer rates up to 300 bps, a connection can have 5 tie trunks in tandem. For data rates of 301–2,400 bps, a connection can have up to 3 tie trunks in tandem. For data rates of 2,401–4,800 bps, a connection can have up to 2 tie trunks in tandem.

Analog tie trunks used in unswitched connections can support up to 9,600 bps.

Digital tie trunks

The digital tie trunk is a high-speed and high-volume trunk interface to a T1 or E1 carrier. It uses a digital signal level 1 (DS1) protocol. By multiplexing 24 64-kbps digital channels onto a single 1.544-Mbps T1 carrier, or 32 64-kbps digital channels onto a single 2.048-Mbps E1 carrier, DS1 offers an economical alternative to the analog tie trunk as well as a high-speed fully digital (without modems) connection between the switches.

The maximum per-channel data rate for DS1 is 64 Kbps, and DS1 trunks can carry voice, voiceband data, or high-speed data communications.

WATS — Wide Area Telecommunications Service

Outgoing WATS service allows calls to certain areas (“WATS bands”) for a flat monthly charge. Incoming WATS trunks allow you to offer toll-free calling to customers and employees.

Applications for different trunk types

To help you select the right type of trunk for a specific application, the following table gives you an overview of key characteristics of different trunk groups. Remember that all analog trunks can carry only voice and voice-grade data.

Type of trunk	Direction	Analog or Digital?	Traffic supported	Transmits digits?
CO, FX, WATS	Incoming Outgoing Two-way	Either	Any kind of voice or data traffic.	No
CPE	N.A.	Either	Any kind of voice or data traffic.	No
DID	Incoming	Either	Only voice and voice-grade data.	Yes
DIOD	Incoming Outgoing Two-way	Either	Any kind of voice or data traffic.	Only for incoming calls
PCOL	Incoming Outgoing Two-way	Analog only	Only voice and voice-grade data.	No
Tie, Access	Incoming Outgoing Two-way	Either	Any kind of voice or data traffic.	Yes

Transmission and supervisory signaling

A trunk is named for its transmission characteristics. For example, trunks are always classified by the direction of the traffic they allow:

- One-way incoming trunk — A local trunk that can be selected (seized) by the far-end connected switch.
- One-way outgoing trunk — A trunk that can be seized by the local switch to call the far-end switch.
- Two-way trunk — A trunk that can be seized by either of the connected switches.

Another transmission characteristic is signaling, which is the transmission of supervision, address, alerting, or other switching information. Supervisory signaling establishes or sets up the connection of the local switch to the distant switch. In general, supervisory signaling has 2 phases:

- Seizure signal — The originating office's signal for a request for service from the distant office.
- Start-dial signal — The distant office's acknowledgment that it is ready to accept dialing from the originating office.

Seizure signals

Ear & Mouth (E&M) supervision

E&M supervision is a symmetric signaling scheme used on private network trunks. DC voltage levels are sent over E&M leads, which are separate from the transmission path. E & M signals indicate on-hook and off-hook states for each end of the connection path.

Ear & Mouth (E&M) supervision on digital trunks

DIOD trunk groups support Continuous and Pulsed E&M Signaling that allows you to make and receive calls over Brazil pulsed or continuous E&M signaling trunks and Hungarian pulsed E&M signaling trunks.

Ground-start (GS) supervision

Ground-start signaling is a supervisory signaling scheme used on public network trunks in which ground is applied on the tip (T) lead by the CO and on the ring (R) lead by the switch. For example, the calling switch on a call to a CO seizes the outgoing trunk by placing a ground on the trunk interface R lead. The CO recognizes the trunk seizure as a request for service and grounds the trunk T lead to indicate to the calling switch that the CO is ready to receive digits. Ground-start signaling is superior to loop-start signaling (described below) for the following reasons:

- The switch can make trunks busy to outgoing calls almost immediately, because the tip ground seizure by the distant switch minimizes the interval during which a two-way trunk can be seized from both ends (called glare).

- Ground-start signaling allows answer supervision, a positive indication that a distant switch has disconnected from a call. Answer supervision has 2 benefits:
  - Callers who remain off-hook after completing a call won't be connected to central office dial tone. For example, this prevents a restricted station from making an unauthorized call after placing an authorized call using the attendant.
  - By holding the trunk busy until the central office disconnects, another switch cannot seize the trunk and be connected to the CO party of the previous call.
- **Loop-start (LS) supervision** — A supervisory signaling scheme used between a telephone and a switch in which the telephone or far-end office completes the current path formed by the trunk wires. The circuit provides one signaling state when it is open and another when it is closed. A third signaling state is achieved by changing the direction or magnitude of current in the loop. The preferred method of loop-start signaling is reverse battery signaling. Disconnect times when using loop-start signaling can result in delays of up to 10–20 seconds. Glare, as described in ground-start, is possible.

  Loop-start signaling does not provide answer supervision. This lack can open a potential for toll fraud in some situations, especially when incoming calls are forwarded off-net. Loop-start is used with analog DIOD trunks, but in general try to use loop start only with one-way trunks.
- **Reverse Battery (RB) supervision** — A supervisory technique on one-way trunks that uses open and closure signals from the originating end and reversals of battery and ground from the terminating end (normally used on direct inward dialing trunks).

Start-dial signals

Network trunks operate as automatic, immediate start, dial tone, wink start, or delay dial according to the type of start-dial signal (alerting) the switch sends out or expects to receive. The different transmissions of alerting are:

- **Automatic** — The originating switch sends no digits or start dial signal and expects the terminating switch to complete the call. The call usually is completed by the attendant or other service such as Centralized Attendant Service (CAS).
- **Immediate start** — The originating switch sends digits immediately without waiting for a start dial signal from the terminating switch.
- **Dial tone** — The terminating switch sends precise dial tone to the originating switch. This indicates that the terminating switch is ready to receive digits.

- **Wink start** — The terminating switch sends a wink start (momentary off-hook) signal to the originating switch. This indicates that the terminating switch is ready to receive digits.
- **Delay dial** — The terminating switch sends a delay dial signal (an off-hook signal followed by an on-hook signal) to the originating switch. This indicates that the terminating switch is ready to receive digits.

Types of address transmission

In addition to seizure and start dial signals, switches have to transmit the digits and characters for telephone numbers. This is called address transmission. Three types of signaling are available:

- **Dial Pulse (DP) addressing** — A method of signaling that consists of regular momentary interruptions of a direct or alternating current at the sending end. The number of interruptions corresponds to the value of a digit or character (alternating current is not used by switches). The interruptions usually are produced by a rotary telephone dial, or may be produced by a sender in a switching system.
- **Dual Tone Multifrequency (DTMF) addressing** — Signaling arrangements (commonly known as touch-tone) that consist of two, simultaneous, dialing signals. One tone is from a low group of four frequencies. The other tone is from a high group of four frequencies. Both tones correspond to digits, letters, or characters (0–9, A–Y, or * and #). One of the tones (1,633 Hz) from the high group is a spare.
- **Multifrequency (MF) addressing** — Signaling arrangements that make use of only 2 frequencies out of 6 to represent 10 decimal digits (0– 9) and 5 auxiliary signals. MF signals are used for called number addressing, calling number identification. They also report whether the far end is ringing or busy.

Analog vs. digital trunks

Analog trunks carry voice and voiceband data communication. “Voice” means that sound of any kind is transformed into electrical waveforms and transmitted within an approximate bandwidth of 300 Hz to 3,400 Hz. “Voiceband data” means that data is transmitted within the voiceband and requires a conversion resource (modem) at both ends of the connection. The data-transmission rate for analog trunks depends on the data-handling capability of the modems in the connection.

Digital trunks represent both sound and data as 0's and 1's and can be configured to carry any kind of voice or data traffic. Digital trunks connect to a DS1 circuit pack and provide a T1 or E1 carrier. DS1 service provides an interface for CO, FX, DID, tie, and WATS trunks. The DS1 interface supports incoming and outgoing dial types of ground-start, loop-start, auto/auto, auto/delay, auto/immed, and auto/wink. Signaling may be robbed-bit or common-channel depending on the trunk type and whether the dial-type is incoming or outgoing. The interface may be used to connect the switch to a toll office directly using wink-start tie trunks for two-way access to the toll network.

Supervision, addressing, and alerting methods have been carried over to digital trunks, which use basically the same signaling scheme as analog trunks when establishing a call. These schemes are handled in a variety of ways to indicate particular calling states, such as on-hook, off-hook, ringing, not ringing, and so on, by using A and B bit-timed signaling. A and B bits carry a 0 or 1 depending on the type of trunk, the near-end channel unit type, far-end channel type, the condition of the trunk (open loop, loop closure, reverse battery, and so on), and whether it is transmit or receive signaling. In addition to the above, refer to ''DS1 Circuit Pack'' on page 728 for trunk-related terms associated with DS1 trunk interfaces.

Interactions

- Brazil Block Collect Call

  In both continuous and pulsed E&M signaling, Block Collect Call is not included.

- Personal Central Office Line (PCOL)

  PCOL trunks cannot use continuous, pulsed, or discontinuous E&M.

Related topics

Refer to the *Avaya MultiVantage™ Hardware Solutions Guide* for information on the types of circuit packs available and their capacities. This manual also lists the maximum number of trunks and trunk groups for each system configuration.

Refer to ''Managing trunks'' on page 403 for administration procedures.

Refer to ''DS1 Trunk Service'' on page 1542 for detailed information on Digital Signal Level 1 trunk service.

Refer to ''ISDN service'' on page 1610 for detailed information about Integrated Services Digital Network trunks.

Uniform Dial Plan

Uniform Dial Plan (UDP) provides a common 3, 4, 5, 6, or 7-digit dial plan - or even a combination of these - that can be shared among a group of switches. The UDP applies interswitch dialing and intraswitch dialing. UDP can be used with an electronic tandem network (ETN); main, satellite, and tributary switches; and Distributed Communications Systems (DCS). In a nutshell, UDP provides extension-to-extension dialing among 2 or more private-switching systems. Note that for DCS, a uniform 4 or 5-digit dialing plan must be used.

Administering UDP

Following is a list of the forms and fields you need to consider when administering UDP

Screen	Field
Dial Plan Analysis Table	■ All
Dial Plan Parameters screen	■ UDP Extension Search Order
Uniform Dial Plan Table	■ All
AAR and ARS Digit Analysis Table	■ All
AAR and ARS Digit Conversion Table	■ All
Node Number Routing	■ Route Pat
Extension Number Portability Numbering Plan	■ All
Route Pattern	■ All

Detailed description

UDP routes calls off the local switch by converting the extension number into a private-network or public-network number. The extension initially dialed by the user can be 3, 4, 5, 6, or 7 digits. The private-network or public-network number is formed by deleting digits from and then inserting digits to the front of the extension. Three types of conversion are supported: AAR, ARS, ENP and EXT.

- For AAR and ARS, the resulting number is analyzed and routed as a private-network (AAR) or public-network (ARS) number. If you wish to block further digit conversion by AAR or ARS, you can enter n(o) in the Conv(ersion) column.

- For ENP, a node number must be entered. Node-number routing uses this number to determine which route pattern should be used to route the call. If the original extension is 4, 5, or 6 digits, an ENP code is chosen based on the first 1 or 2 digits of the dialed string. Because it is not used for routing, the ENP code can be independent of location.
- If EXT is entered, the resulting number (which may be unchanged from the dialed extensions) is routed over again as an extension. If you wish to block further digit conversion by UDP, you can enter n(o) in the Conv(ersion) column.
- If no UDP entry exists for a particular extension or range of extensions, those extensions are considered local.

UDP conversion is specified for individual extensions or groups of extensions sharing the same leading digits, via the UDP screen and the matching pattern. A matching pattern is the desired leading digits of an extension. For example, with a 5-digit UDP “123” is the matching pattern for all 100 possible 5-digit extensions beginning with “123”. By contrast, “12345” would be a matching pattern specifying a single extension.

Each extension code can be assigned to 1 of 6 possible treatments.

- UDPCode — Conversion to AAR with given location code, further conversion suppressed
- AARCode — Conversion to AAR with given location code, further conversion allowed
- ENPCode — Conversion to private network number (via ENP screen), route to given node number routing
- TempOOS — Temporarily out of service, give reorder
- Local — Local range of extensions
- blank — Similar to local; however, this extension is not chosen when the system is asked to “add a station”

When a user at a switch that is included in a UDP dials an extension, Avaya MultiVantage first checks to see if the extension is assigned to a local station on that switch. If so, the call is routed to the station, and UDP is not invoked. If the extension is not found locally, the system checks to see if the extension matches an assigned extension code. If the extension matches an assigned extension code, the system performs the specified conversion into a private network number and routes the call as specified. If more than one extension code matches, the “best” match (most explicit digits) is used. For example, 1234 is chosen over 123 if 12345 is dialed. However, 123 is chosen if 12355 is dialed. If no matching extension code is found, the user receives intercept treatment. Or, if Extended Trunk Access (ETA) is enabled, the call routes via ETA.

Example of how UDP works

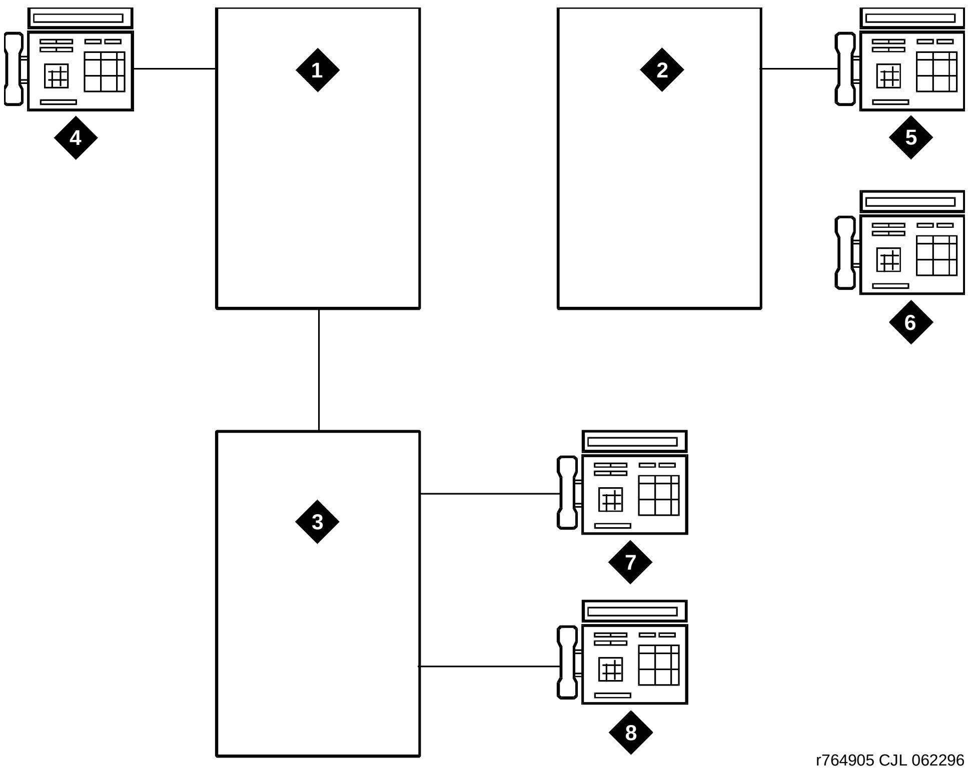
To administer, assign each UDP code:

- To a private network location code (RNX) or node number. (The RNX is equivalent to an office code of a central office in a public network. This RNX determines how a UDP call is routed.)
- Assign each UDP code as either local or remote to the switch.

To understand the function of a UDP, review the following example.

A given station is called using the same 5-digit extension regardless of where in the ETN the call originates. Three switches are included in the example. Each has been assigned a list of RNX and UDP codes. See Figure 64 on page 1795.

SWITCH	RNX	UDP CODE
A	224	41
C	223	51
C	223	52
B	222	60
B	222	61



1. Switch A
   Dial Plan for extensions is 41
   RNX=224
2. Switch B:
   Dial Plan for Extensions is 60
   and 61 RNX=222
3. Switch C:
   Dial Plan for extensions is 51
   and 52 RNX=223
4. Extension 41000
5. Extension 61234
6. Extension 60123
7. Extension 51234

Figure 64. UDP Example

If a user at extension 41000 wants to call extension 61234, there are two choices:

- Dial 61234
- Dial the AAR access code followed by 222-1234

If 61234 is dialed, the system recognizes 61 as a remote UDP, determines the associated RNX (222), and uses AAR to route the call to 222-1234.

If the AAR access code and 222-1234 are dialed, the system finds the route pattern for RNX 222 and routes the call to the switch associated with that RNX.

Whenever UDP is used to route a call to another switch, the correct digit deletion and insertion must be specified within the route pattern so that the receiving switch gets digits in the format it expects. Avaya MultiVantage can be configured in several different ways.

- If AAR is available on the receiving switch, Subnet Trunking can be used to insert the AAR feature access code on the originating switch or digit insertion may be used to insert it on the receiving switch. The receiving switch then uses AAR digit conversion to convert 222 with 7 digits to an extension by deleting 3 digits and inserting a 6.
- If AAR is not available on the receiving switch, Subnet Trunking must be used on the originating switch to delete the 222 and insert the digit 6 at the beginning of the extension number so that the receiving switch can continue to route correctly.

If the user at extension 51234 on Switch C dials extension 61234, the call must first go through Switch A before proceeding to Switch B. When 61234 is dialed, the system recognizes 61 as a UDP code, determines the associated RNX (222), and uses AAR to route the call. The AAR feature access code plus 222-1234 are outpulsed to Switch A. Switch A then recognizes the RNX 222 as a remote switch and routes the call to Switch B and extension 61234. This same type of call routing occurs when an extension at Switch B calls an extension at Switch C.

If extension 61234 on Switch B calls extension 61235, the system recognizes 61 as a local UDP code and routes the call directly to extension 61235.

Considerations

- In North American network environments, extensions beginning with 0 may route to an attendant. You are encouraged to use another number as the leading digit when assigning extensions.
- When you call an extension on another switch, there may be a slight delay before call-progress tones are applied. This delay is due to the trunk signaling necessary to complete the call to the remote switch.
- When you select the option to look at the UDP table first, calls that might otherwise terminate at a local extension route over the network. This is easily reversed. When you remove the extensions from the UDP table, the local extension can be dialed again.

- If AAR is active, FRLs and Traveling Class Marks (TCMs) can be sent along with the private network number. UDPCode and AARCode conversions use the FRL assigned to the caller. ENPNode conversion always raises the FRL to the maximum (7).

  If an FRL is insufficient to access the facility, access is denied. There is no prompt for an authorization code even if authorization codes are enabled and administered.

- If AAR is not active, tandem-tie trunks should not be used to transport UDP numbers. The TCM is not recognized at the terminating switch.

NOTE:
Never use tandem-tie trunks to transport UDP numbers (the TCM and Hop count will not be recognized as such following the extension received at the receiving switch).

Interactions

- Automatic Alternate Routing

  AAR routes UDP calls. (Included with UDP is the required AAR subset.) If AAR is enabled in addition to UDP, then the 7-digit AAR number provides the same routing as UDP.

- Dial Plan

  — All of the extension numbers on a switch are not necessarily part of UDP. Any that do not belong to UDP are handled by a regular, non-UDP Dial Plan associated with the switch.

  — When administering the Dial Plan screen and designating a group of extensions as UDP non-local, you can specify whether you want to search for local extensions first or last. This allows flexibility to change extensions from local to non-local and vice-versa.

- DID Trunk Group

  DID calls to a UDP extension number may require that the DID trunk group insert enough digits to create the full extension number. For example, if the DID trunk provides 4 digits, but a 5-digit UDP is in place, then the appropriate leading digit must be inserted.

- Distributed Communications System

  UDP is required when DCS is provided. The necessary UDP software is provided with the DCS software.

- Extension Number Portability

If an extension code is administered to use ENP node routing, ENP routes the call to the correct switch. If AAR is enabled in addition to UDP, the 7-digit AAR number provides the same routing as UDP (that is, via ENP).

Voice Message Retrieval

Voice Message Retrieval allows attendants, phone users, and remote-access users to retrieve Leave Word Calling (LWC) and Call Coverage messages.

Detailed description

Voice Message Retrieval is only used for the retrieval of messages. When a terminal is in Voice Message Retrieval mode, it cannot be used to make calls or access other features. Voice Message Retrieval can be used to retrieve your own messages or messages for another user. However, a different user’s messages can only be retrieved by a user at a phone or attendant console in the coverage path, by an administered system-wide message retriever, or by a remote-access user when the extension and associated security code are known. The number of simultaneous Voice Message Retrieval users possible depends on the number of speech-synthesizer circuit packs used in the system.

Certain phones and attendants can be designated for system-wide message retrieval. These system-wide retrievers are the same as those used for Display Message Retrieval and have the same privileges. Voice Message Retrieval cannot be accessed from rotary phones.

You can use the system to restrict unauthorized users from retrieving messages. Use the Lock function to restrict a phone and the Unlock function to release the restriction. Users activate Lock by dialing a system-wide access code. They cancel Lock by dialing a system-wide access code and then an Unlock security code unique to the phone. These functions only apply to the phone where the function is active. The system-wide access codes and security code used for the Lock and Unlock functions are the same as those used for LWC message retrieval by display. You can assign a status lamp to show the lock status of the phone.

Interactions

- AUDIX Interface

  Retrieval of LWC messages via Voice Message Retrieval is separate and distinct from retrieval of messages via INTUITY. LWC messages left for a principal on AUDIX cannot be accessed via Voice Message Retrieval. However, the caller of Voice Message Retrieval is informed of any new messages for the principal on AUDIX:

— The Voice Message Retrieval voices that there are AUDIX messages.

— The Display Message Retrieval displays “Message Center AUDIX Call.”

If your system has a voice-synthesizer circuit pack and LWC Activation is active, users can retrieve messages from two locations:

— Users can retrieve LWC messages with Voice Message Retrieval.

— Users can retrieve all other messages with AUDIX.

If you do not have a TN725B speech-synthesizer board, non-display phone users cannot retrieve LWC messages sent via the LWC button on a phone.

- Bridged Call Appearance

  Voice Message Retrieval on a Bridged Call Appearance functions the same as if it were activated by the primary extension associated with the bridged call appearance.

- Leave Word Calling

  Voice Message Retrieval enhances LWC by allowing any authorized touch-tone phone user to retrieve messages.

Related topics

Refer to ‘‘Audible Message Waiting’’ on page 1362 for information about letting users know they have messages.

Refer to ‘‘Feature Access Code’’ on page 752 for information about and field descriptions on the Feature Access Code screen. Complete the LWC Message Retrieval Lock, LWC Message Retrieval Unlock, Voice Coverage Message Retrieval Access Code, and Voice Principal Message Retrieval Access Code fields on this screen to administer voice message retrieval.

Refer to ‘‘Feature-Related System Parameters’’ on page 766 for information about and field descriptions on the Feature-Related System Parameters screen. Complete the Stations With System-Wide Retrieval Permission and Message Waiting Lamp Indicates Status For fields on this screen to administer voice message retrieval.

Refer to ‘‘Station’’ on page 1066 for information about and field descriptions on the Station screen. Complete the Security Code field on this screen to administer voice message retrieval.

Voice Messaging Systems

Avaya MultiVantage supports several Avaya voice or multimedia messaging systems. These systems allow users to send, retrieve, store, and forward messages, as well as perform many other tasks associated with messages. In addition to supporting multiple AUDIX systems, MultiVantage can have multiple hunt groups associated with a single AUDIX system. This allows partitioning of the voice ports into different hunt groups and different coverage paths to cover different voice ports. Thus voice ports can be reserved for particular users or groups of users (for example, those that use unique coverage paths).

The following features do not use coverage paths:

- Transfer into AUDIX with the feature access code or the GOTO COVER button
- Last Call

If a local AUDIX and a remote AUDIX use the same hunt-group numbers, calls route to the local hunt group.

Avaya MultiVantage supports the following systems:

INTUITY AUDIX

INTUITY AUDIX runs on a separate MAP/5, MAP/40, or MAP/100 PC. The switch communicates with INTUITY AUDIX via analog voice ports and a data link. The switch can also communicate with INTUITY AUDIX without the data link. In this case, the switch and INTUITY AUDIX communicate by sending and receiving special strings of touch-tone codes (dual tone multifrequency tones) via analog voice ports. These touch-tone codes are called *mode codes* and carry data such as calling party ID, called party ID, and type of call.

INTUITY AUDIX allows up to 64 ports. This means up to 64 people can be simultaneously retrieving or leaving messages. INTUITY AUDIX also supports fax and e-mail messaging.

For more information, refer to *INTUITY AUDIX System Description* or *INTUITY AUDIX Administration*, and *INTUITY Messaging Solutions Integration with System 75, Generic 1 and 3, and R5/6*.

Embedded AUDIX	(Not available with GuestWorks) Embedded AUDIX runs on a multifunction circuit pack assembly. This assembly fits into 2 contiguous slots in the MultiVantage switch. Embedded AUDIX communicates with the switch via analog voice ports with a data link. Embedded AUDIX can also communicate exclusively via analog voice ports when set up to emulate a digital phone set. Embedded AUDIX allows up to 16 ports. For more information, refer to *DEFINITY AUDIX System Feature Descriptions, DEFINITY AUDIX System Administration, Switch Administration for DEFINITY AUDIX System*, or *DEFINITY AUDIX System Forms Reference*.
Octel Serenade	Octel Serenade is a voice messaging system that supports MultiVantage systems via QSIG signaling protocols.
Octel 100	Octel 100 runs on personal computer running the OS2 operating system.

Other non-Avaya messaging systems may also use mode codes to work with Avaya MultiVantage.

Centralized Voice Mail

You can use a single voice mail system to support multiple MultiVantage systems in a DCS network. In addition, you can use a voice mail system to support multiple MultiVantage systems and Merlin Legend in a network via mode code. For more information, see *Administration for Network Connectivity for Avaya MultiVantage™ Software*.

Security Measures

SECURITY ALERT:

Fraudulent long-distance calls can be placed through INTUITY AUDIX, embedded AUDIX, and AUDIX R1 if proper security precautions are not implemented.

The following actions help secure your system from unauthorized use.

- For Intuity AUDIX, embedded AUDIX, and AUDIX R1
  - Remove any unused or unassigned mail to help prevent exchange of information through unassigned Voice Mail.
  - Secure system lines that serve AUDIX and control where calls can be placed.

- Assign a restrictive Class of Restriction (COR), Class of Service (COS), and Facilities Restriction Level (FRL) to the station lines and trunks serving AUDIX.
- Use switch Call Detail Recording (CDR) reports to determine if the lines are being used for calls that are normally not within your sphere of business.
- Change default passwords on voice mailboxes immediately after installation and use random numbers for passwords.
- Require passwords with at least 5 digits.
- Change system administration passwords to alphanumeric codes.
- INTUITY AUDIX, AUDIX R1, and embedded AUDIX provide a maintenance/administration port. A remote port security device provides an added layer of security to prevent unauthorized access to this port.

■ For INTUITY AUDIX and embedded AUDIX

- Determine whether to only allow transfers to other AUDIX subscribers or to any extension of the correct length. The most secure approach is to only allow transfers to other AUDIX subscribers. If you decide to allow transfers to any extension, administer the COR on the AUDIX ports to prevent calls outside the PBX or immediate Distributed Communications System (DCS) site network.

■ For AUDIX only

- Activate Enhanced Call Transfer (ECT) to help prevent having billable calls placed from unauthorized transfers outside of the system. ECT performs call transfer over the data link between AUDIX and the Avaya MultiVantage. (The destination extensions must be administered on the switch dial plan.)

Interactions

■ Attendant Conference

An attendant who has split a call can conference the call with AUDIX by dialing the Transfer Into AUDIX access code. The attendant presses Release to drop out of the conference call.

■ Automatic Call Distribution (external AUDIX only)

You can administer a hunt group as an AUDIX ACD split. You can then obtain AUDIX traffic measurements with the ACD Call Management System. Login occurs when AUDIX signals the switch to make a voice port available for AUDIX service. Logout occurs when AUDIX signals the switch to disable the port.

AUDIX and ACD CMS must be connected to the same switch. If AUDIX in the DCS feature is active, a CMS located on a switch other than the host switch (AUDIX location) does not provide measurements for the AUDIX ports.

Because AUDIX frequently takes voice ports in and out of service for maintenance testing, high login activity may be seen for the AUDIX split in the measurement reports.

On CMS reports that display an agent’s login ID, AUDIX voice ports always show a login ID that is the same as the extension, even if you have not administered login IDs on the switch.

- Call Coverage

  When a coverage call successfully completes to AUDIX or routes from a remote switch to the host switch because of coverage, the principal is dropped from the call (no temporary bridge appearance is maintained).

  Coverage calls from a remote switch that fail to reach AUDIX as a coverage point cannot be returned to the original coverage path on the remote switch.

- Call Forwarding

  An AUDIX user can forward calls to a remote AUDIX hunt group or to the host AUDIX hunt group. You must correctly administer the AUDIX destination for the remote AUDIX hunt group.

- Call Monitoring

  Call Monitoring allows users to pick up the handset after the call goes to AUDIX to listen to the message being left by the caller. This allows the user to determine whether they want to talk to the caller or let them complete their message.

- Call Transfer

  A call transfer out of AUDIX can be to a UDP extension. If the UDP extension is on a remote switch, the call is treated as a direct call. Additional trunks are used for calls transferred between DCS nodes.

  Calls may be transferred into AUDIX by users handling redirected calls for principals who are AUDIX subscribers.

- Class of Restriction

A high FRL assigned in the COR allows callers to transfer to long-distance numbers.

To prevent toll fraud, assign a low FRL to the AUDIX system ports.

- DCS — Leave Word Calling (external only)

  In a DCS network, the called party may be on a different switch than the calling party. If the DCS link is down, attempts to store Leave Word Cancel messages are denied and intercept tone is returned. LWC requests are always denied for principals with AUDIX LWC; in some instances, the request to cancel LWC may appear to be active when it actually is not.

  When the local switch communicates with INTUITY AUDIX via mode codes, INTUITY AUDIX cannot support remote DCS switches.

- Facility Test Call

  Unauthorized calls can be placed using the facility test-call access code.

  To prevent toll fraud, remove the facility test call access code.

- Leave Word Calling

  You can have a principal’s LWC messages kept by AUDIX. The principal can retrieve a message by calling AUDIX. The principal cannot retrieve the message using other retrieval methods, but is notified of its existence by AUDIX.

  For other messaging services, you can have AUDIX report the existence of waiting LWC messages for the principal, but not the message content. The principal can retrieve the message using other retrieval methods, but is still notified of the existence of AUDIX messages.

  If the data link between the system and AUDIX is down, attempts to activate LWC for an AUDIX-covered principal are denied and reorder tone is returned.

  If a caller attempts to cancel a LWC message sent to AUDIX, the caller receives intercept tone if the called party is on the same switch. The caller receives confirmation tone if the called party is on another switch in the DCS network as long as the DCS data link to the called party’s switch is operational, *even though the message is not actually canceled*.

  When the local switch communicates with INTUITY AUDIX via mode codes, INTUITY AUDIX does not accept or store LWC messages in user mailboxes.

- Ringback Queueing

Ringback Queueing does not apply to AUDIX calls. On direct calls to a remote AUDIX, if all trunks to the host AUDIX are busy, busy tone is returned. On coverage calls, if all trunks are busy, AUDIX is treated as a busy coverage point. If there are coverage points after AUDIX, then the call terminates at those points.

- Single-Digit Dialing and Mixed-Station Numbering

  AUDIX is designed for use with a Uniform Dial Plan. It supports only one extension length (3-, 4-, or 5-digit) that is used by AUDIX subscribers. Single-Digit and Mixed Station Numbering cannot be used. However, nothing prohibits connecting a switch to AUDIX that provides these features, as long as all AUDIX subscribers have the same extension length.

- Temporary Bridged Appearance

  Stations that normally have a temporary bridged appearance with their coverage point do not have that appearance if the coverage point is AUDIX.

Related topics

Refer to ''Audible Message Waiting'' on page 1362 for information about letting users know they have messages.

Refer to ''Voice Message Retrieval'' on page 1798 for information about retrieving messages.

Refer to *INTUITY Integration with System 75 and DEFINITY Communications System* for the procedures on how to administer the switch for voice messaging for Intuity AUDIX.

Refer to *Switch Administration for the embedded AUDIX System* for the procedures on how to administer the switch for voice messaging for embedded AUDIX.

Refer to *Switch Administration for AUDIX Voice Messaging* for the procedures on how to administer the switch for voice messaging for AUDIX R1.

Refer to ''Data modules'' on page 679 for information about and field descriptions on the Data Modules screen. Complete this screen (for one AUDIX link) to administer the switch for voice messaging.

Refer to ''Packet Gateway Board'' on page 1018 for information about and field descriptions on the Packet Gateway Board screen. Complete all fields on this screen to administer the switch for voice messaging.

Refer to ''Mode Code Related System Parameters'' on page 983 for information about and field descriptions on the Mode Code Related System Parameters screen. Complete all fields on this screen to administer the switch for voice messaging.

Refer to ''Hunt Group'' on page 840 for information about and field descriptions on the Hunt Group screen. Complete all fields (as required) on this screen to administer the switch for voice messaging.

Refer to ''Class of Service'' on page 650 for information about and field descriptions on the Class of Service screen. Complete the Call Fwd-All Calls field on this screen to administer the switch for voice messaging.

Refer to ''Coverage Path'' on page 672 for information about and field descriptions on the Coverage Path screen. Complete all fields on this screen to administer the switch for voice messaging.

Refer to ''Trunk Group'' on page 1167 for information about and field descriptions on the Trunk Group screen. Complete all fields on this screen to administer voice messaging for a DCS configuration.

Whisper paging

Whisper paging allows one user to interrupt or “barge in” on another user’s call and make an announcement. The paging user dials a feature access code or presses a feature button, then dials the extension they want to call.

Only the person on the paged extension can hear the page: other parties on the call cannot hear it, and the person making the page cannot hear anyone on the call. If the paged user has a display phone, he or she can see who is making the whisper page.

For example, let’s say users A and B are on a call. C has an urgent message for A and makes a whisper page. All 3 users hear the tone that signals the page, but only A hears the page itself. The person making the page, C, cannot hear A or B.

Brief description

Allowing users to make whisper pages

To make a whisper page users dial a feature access code or press a feature button, then dial the extension of the user they are trying to reach.

- To assign a feature access code, enter a code in the Whisper Page Activation Access Code field on the Feature Access Code screen.
- To give users a feature button for making a whisper page, use the Station screen and administer a Whisper Page Activation button on users' phones.

Allowing users to answer whisper pages quickly

To give users a feature button for answering a whisper page, use the Station screen and administer an Answerback button on users' phones.

NOTE:
You cannot administer an Answerback button on an attendant console. Attendants can make whisper pages but cannot receive them.

Normally, before a paged user can answer a whisper page he or she must complete the active call or put it on hold. However, you can give users the ability to put an active call on hold and speak directly to the person making a whisper page simply by pushing a feature button. Once the Answerback button is pressed, the user can treat both the paging call and the original call as separate calls and all call-related features (conference, transfer, etc.) operate normally.

Allowing users to block whisper pages

To give users a feature button to block incoming whisper pages, use the Station screen and administer a Whisper Page Off button on users' phones.

Administer this function on a feature button with a lamp so that users can tell whether or not blocking is active. Users can activate blocking even when they're on a call.

NOTE:
You cannot administer a Whisper Page Off button on a soft key.

Two features, Do Not Disturb and Privacy — Attendant Lockout, also block incoming whisper pages.

Call redirection overrides

If a paged user is not on an active call, a whisper page is converted to a priority call that overrides any of the following call redirection features:

- Call Forward All Calls
- Call Forward Busy
- Call Forward Don't Answer
- Send All Calls
- Go To Cover
- Call Coverage

For example, let's say Call Forwarding — All Calls is activated on a phone. If there are no active calls, a whisper page to that phone will ring at that phone as a priority call. For information on blocking incoming whisper pages, refer to ''Allowing users to block whisper pages''.

Group answering environments

Whisper paging does not work with extensions assigned to a group answering environment. You cannot place a whisper page to the main extension assigned to a hunt group, split, skill, or terminating extension group. You cannot place a whisper page to any extension that is a member of one of these groups.

Network restrictions

Whisper paging does not work across networks (such as Distributed Communication System networks or Electronic Tandem Networks): both the paging user and paged user must be on the same switch.

Speakerphones

When a call is on the speaker, an incoming whisper page is heard over the speaker too. When the group listening feature on a 6400-series phone is active, an incoming whisper page is heard on both the handset and the speaker.

Interactions

- Attendant

  Attendants cannot intrude on a whisper page. If an attendant is using intrusion to talk to a user, that user cannot receive a whisper page.

  An attendant may initiate a whisper page while a call is split away using auto-manual splitting. However, they cannot use Release, Hold, or Split after the page is made.

- Bridged Call Appearances

  Whisper pages are designed to reach a specific user associated with a specific extension.

  - When an extension and one or more of its bridged appearances are in use, parties on the bridged appearances hear the tone that signals an incoming whisper page but only the user on the principle extension hears the announcement. Only the display on the principle extension shows the whisper page message.
  - When all appearances are idle or only a bridged appearance is in use, a whisper page rings the principal extension with priority ringing.
  - If a user makes a whisper page on a call appearance that is a member of a bridge group, then no others users in the bridge group can connect to the call while the whisper page intrusion is active.

- Busy Verification

  You can't make a whisper page to an extension while it's being busy-verified. You can't busy-verify an extension while it's making or receiving a whisper page.

- Calling Restrictions — Origination

  Phones with this restriction cannot make whisper pages.

- Calling Restrictions — Termination

  Phones with this restriction cannot receive whisper pages.

- Class of Restriction (COR)

  A station user must have a COR that allows for station-to-station calling in order to perform Whisper Paging to a member outside of their own COR. Calling and called party restrictions also determine which extensions can make and receive whisper pages.

- Conference

  Everyone on a conference call hears the tone that signals an incoming whisper page. But only the owner of the paged extension hears the page, and only the display on that phone shows the whisper page message.

  If a conference call already has the maximum number of parties and trunks allowed, you cannot make a whisper page to any of the participants. Parties cannot be added to a conference call if an active whisper page is on the call.

- Data Privacy — Permanent or Temporary

  Any station that has Data Privacy activated cannot make a whisper page.

- Data Restriction

  A whisper page to a station is denied when Data Restriction is enabled on a station or trunk.

- Expert Agent Selection

  You can't make a whisper page by dialing an agent's Logical Agent ID. Pages must be made to a physical extension.

- Go to Cover

  If you make a whisper page and then press your Go To Cover button while the page is in progress, the Go To Cover button doesn't work. The opposite is true as well. If you activate Go to Cover and then press the whisper page activation button, you aren't able to make a whisper page.

- Last Number Dialed

  When you make a whisper page, it's tracked as the last number dialed.

- QSIG

  This feature does not operate in a QSIG environment.

- Remote access

  You can't make a whisper page by remote access. Both the paging party and paged party must be on the same switch or the attempt is denied.

- Tenant Partitioning

  Whisper paging is permitted across tenant partitions if the assigned classes of restriction allow for intercom calling between members of different partitions. This feature is especially useful to attendants who serve multiple partitions.

  NOTE:
  It is important for system administrators to ensure that this feature is managed appropriately in systems with tenant partitioning. Some tenants may not want other tenants to interrupt their calls.

- Transfer

  A call cannot be transferred during an active whisper page.

- Vector Directory Number (VDN)

  You can't make a whisper page to a VDN. Pages must be made to a physical extension.

Related topics

Refer to ''Paging users who are on active calls'' on page 471 to administer whisper paging.

Wideband Switching

(Not available with Offer B) Wideband Switching provides the ability to dedicate 2 or more ISDN-PRI B-channels or DS0 endpoints for applications that require large bandwidth. It provides high-speed end-to-end connectivity between endpoints where dedicated facilities are not economic or appropriate. ISDN-BRI trunks do not support wideband switching.

Brief description

ISDN-PRI (and the emulation of it by Asynchronous Transfer Mode-Circuit Emulation Service (ATM-CES)) divides a T1 or E1 trunk into 24 (31 for E1) information channels and one signaling channel for standard narrowband communication. Certain applications, like video conferencing, require greater bandwidth. You can combine several narrowband channels into one wideband channel to accommodate the extra bandwidth requirement. Avaya MultiVantage serves as a gateway to many types of high-bandwidth traffic. Wideband switching is also supported by the Expansion Interface (EI) circuit pack and the DS1 Converter circuit pack (for Center Stage Switching or directly connected port networks) and the ATM-EI circuit pack (for Asynchronous Transfer Mode connected port networks). ATM-CES supports wideband switching only for access, tie, and tandem trunks, not for line-side connections.

Wideband Switching supports:

- High-speed video conferencing
- WAN disaster recovery
- Scheduled batch processing (for example, nightly file transfers)
- LAN interconnections and imaging
- Other high bandwidth applications involving high-speed data transmission, video transmission, etc.

How to administer Wideband Switching

The following list shows the required screens you must to set up Wideband Switching:

- Access Endpoint
- PRI Endpoint
- ISDN trunk group
- Route Pattern
- Fiber Link Administration (Optional: refer to the DEFINTY services documentation for information about this screen.)

Channel allocation

For standard narrowband communication, ISDN-PRI divides a T1 or E1 trunk as follows:

- T1 trunks are divided into 23 information channels and 1 signaling channel
- E1 trunks are divided into 30 information channels, 1 signaling channel, and 1 framing channel

Certain applications, like video conferencing, require greater bandwidth. You can combine several narrowband channels into one wideband channel to accommodate the extra bandwidth requirement. Avaya MultiVantage serves as a gateway to many types of high-bandwidth traffic. In addition, DS1 converters are used for wideband switching at remote locations.

The following table provides information on Wideband Switching channel types.

Channel Type	Number of Channels	Data Rate
H0	6	384 Kbps
H11	24	1536 Kbps
H12	30	1920 Kbps
NXDS0 (T1)	2-24	128–1536 Kbps
NXDS0 (E1)	2-31	128–1984 Kbps

Perform wideband line-side channel allocation using one of three allocation algorithms: fixed, flexible, or floating.

- Fixed allocation — Provides contiguous-channel aggregation. The starting channel is constrained to a predetermined starting point. (Used only for H0, H11, and H12 calls.)
- Flexible allocation — Allows a wideband call to occupy noncontiguous positions within a single T1 or E1 facility.
- Floating allocation — Enforces contiguous-channel aggregation. The starting channel is not constrained to a predetermined starting point.

Typical uses

A typical video application uses an ISDN-PRI interface to DS0 1 through 6 of the line-side facility. Refer to the following figure.

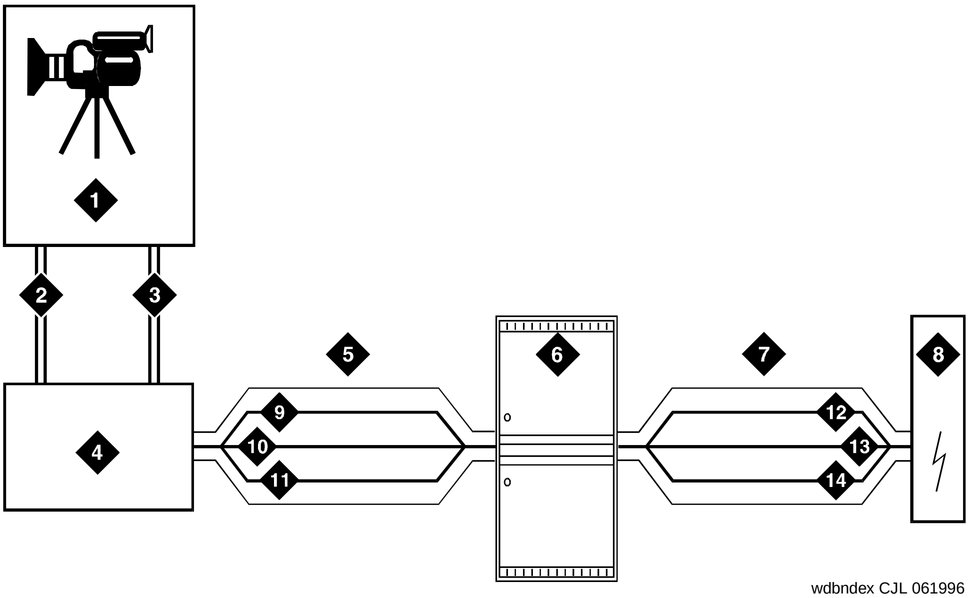


Figure Notes

1. Video application
2. Port 1
3. Port 2
4. ISDN terminal adaptor
5. Line-side ISDN-PRI
6. Avaya Media Server
7. ISDN or ATM-CES trunk
8. Network
9. DS0 24 D-channel
10. DS0 23 unused
11. DS0 1–6 wideband
12. DS0 24 D-channel
13. DS0 7–23 narrow bands
14. DS0 1–6 wideband

Figure 65. Typical video broadband application

Endpoint applications with signaling

An endpoint application is the origination or destination point of a wideband call. Endpoint applications can be any number of data applications based on the customer's particular needs.

ISDN-PRI terminal adapters

For wideband switching with non-ISDN-PRI equipment, you can use an ISDN-PRI terminal adapter. ISDN-PRI terminal adapters translate standard ISDN signaling into a form that can be used by the endpoint application and vice versa. The terminal adapter also must adhere to the PRI-endpoint boundaries as administered on the Avaya MultiVantage switch when handling both incoming (to the endpoint) applications and outgoing calls.

The terminal adapter passes calls to and receives calls from the line-side ISDN-SETUP messages indicating the data rate and specific B-channels (DS0) to be used and communicates all other call status information via standard ISDN messages. Refer to *DEFINITY Line-Side ISDN Primary Rate Interface Technical Reference* for more information.

Line-side (T1 or E1) ISDN-PRI facility

A line-side ISDN-PRI (T1 or E1) facility is comprised of a group of DS0s (24 for a T1 facility and 32 for an E1 facility). In this context, these DS0s are also called channels. T1 facilities have 23 B-channels and a single D-channel. E1 facilities have 30 B-channels, 1 D-channel, and a framing channel. Data flows bi-directionally across the facility between the switch and the ISDN-PRI terminal adapter.

PRI-endpoints

A PRI-endpoint (PE) is a combination of DS0 B-channels on a line-side ISDN-PRI facility that has been assigned an extension.

A PRI-endpoint can support calls of lower bandwidth. In other words, a PE having a width 6 (six DS0s) can handle a call of one channel (64 Kbps) up to and including six channels (384 Kbps). Also, a PE can support calls on non-adjacent channels. For example, an endpoint application connected to a PE defined as using B-channels 1 through 6 of an ISDN-PRI facility could originate a call using B-channels 1, 3, and 5 successfully. If the PE has been administered to use flexible channel allocation, the algorithm for offering a call to the PE starts from the first DS0 administered to the PE. Since only one active call is permitted on a PE, contiguous B-channels are always selected unless one or more B-channels are not in service.

A PE remains in service unless all of its B-channels are out of service. In other words, if B-channel 1 is out of service and the PE is five B-channels wide, the PE could still handle a wideband call of up to four B-channels in width. A PE can only be active on a single call at any given time; that is, it is either considered idle, active (busy), or out of service.

One facility can support multiple separate and distinct PRI-endpoints (several extensions) within a single facility. Non-overlapping contiguous sets of DS0s (B-channels) are associated with each PE.

Universal digital signal level 1 board

The UDS1 board is the interface for line-side and network facilities carrying wideband calls.

Non-signaling endpoint applications

Wideband can also support configurations using non-signaling (non-ISDN-PRI) line-side T1 or E1 facilities. The endpoint applications are the same as those defined for configurations with signaling.

Data service unit/channel service unit

This unit simply passes the call to the endpoint application. Unlike terminal adapters, the DSU/CSU does not have signaling capability.

NOTE:
No DSU/CSU is needed if the endpoint application has a fractional T1 interface.

Line-side (T1 or E1) facility

This facility, like the ISDN-PRI facility, is composed of a group of DS0s (23 for a T1 facility and 30 for an E1 facility). Line-side facilities are controlled solely from the switch. Through the access-endpoint command, a specific DS0 or group of DS0s is assigned an extension. This individual DS0 or group, along with the extension, is known as a wideband access endpoint (WAE).

Wideband access endpoint

WAEs have no signaling interface to the switch. These endpoints simply transmit and receive wideband data when the connection is active.

NOTE:
The switch can determine if the connection is active, but this does not necessarily mean that data is actually coming across the connection.

A WAE is treated as a single endpoint and can support only one call. If all DS0s comprising a wideband access endpoint are in service, then the wideband access endpoint is considered in service. Otherwise, the wideband access endpoint is considered out of service. If an in-service wideband access endpoint has no active calls on its DS0s, it is considered idle. Otherwise, the wideband access endpoint is considered busy.

Multiple WAEs are separate and distinct within the facility and endpoint applications must be administered to send and receive the correct data rate over the correct DS0s. An incoming call at the incorrect data rate is blocked.

Guidelines and examples

This section examines wideband and its components in relation to the following specific customer usage scenarios:

- High-speed video conferencing
- Data backup connection
- Scheduled batch processing
- Primary data connectivity
- Networking

High-speed video conferencing

All data rates are multiples of 64 Kbps; from 128 Kbps to 1,536 Kbps (T1) and 1,984 Kbps (E1) are supported. Key customer data rates are listed below.

Channel Type	Number of Channels	Data Rate
H0	6	384 Kbps
H11	24	1536 Kbps
H12	30	1920 Kbps
NXDS0 (T1)	2-24	128–1536 Kbps
NXDS0 (E1)	2-31	128–1984 Kbps

Data backup connection

Using wideband for data transmission backup provides customers with alternate transmission paths for critical data in the event of primary transmission path failure.

Scheduled batch processing. Scheduled batch processing applications are used for periodic database updates (for example, retail inventory) or distributions (for example, airline fare schedules). These updates are primarily done after business hours and are often referred to as nightly file transfers. Wideband meets the high bandwidth requirements at low cost for scheduled batch processing. In addition, wideband allows the dedicated-access bandwidth for busy-hour switch traffic to be used for these applications after business hours; no additional bandwidth costs are incurred.

The non-ISDN backup data connection is also appropriate for scheduled batch processing applications. Administered Connections are used to schedule daily or weekly sessions originating from this application.

Primary data connectivity. Permanent data connections (those always active during business hours), such as interconnections between local area networks (LANs), are well suited for Avaya MultiVantage when ISDN-PRI endpoints are used. The ISDN end-to-end monitoring and the endpoint's ability to react to failures provide for critical data availability needs. With ISDN, endpoints can detect network failures and initiate backup connections through the switch; ISDN endpoints can also establish additional calls when extra bandwidth is needed.

Any failures not automatically restored by Avaya MultiVantage are signaled to the endpoint application, which can initiate backup data connections over the same PRI endpoint. Avaya MultiVantage routes the backup data connections over alternate facilities if necessary.

Networking

All of the wideband networking is over ISDN-PRI facilities (and the emulation of them by ATM-CES) but may connect to a variety of networks, other domestic interexchange carriers' services, private line, RBOC services, and services in other countries.

ISDN-PRI trunk groups and channel allocation

Only ISDN-PRI trunks (and the emulation of them by ATM-CES) support wideband calls to the network. Wideband's bandwidth requirements have necessitated modification of the algorithms by which trunks look for clear channels. The following section describes the search methods and their relationship to the available wideband data services.

Facility lists

A wideband call accessing the network must reside on a single ISDN-PRI facility. Trunks within a trunk group must be organized based on the facility on which they reside. This is accomplished by compiling a facility list as trunks are administered to a trunk group; if a trunk is added to a trunk group from a facility not already on that trunk group's list, that facility is added to the list in an order based on the facility's signaling group number and interface identifier. In other words, the facility list is compiled in an ascending order based first on signaling group number and second on the interface identifier assigned to the facility within the signaling group. For example, if three facilities having signaling group/interface identifier combinations of 1/1, 1/2, and 2/1 were associated with a trunk group, then a call offered to that trunk group would search those facilities in the order as they were just listed. Also note that since trunks within a given facility can span several trunk groups, a single facility can be associated with several different trunk groups.

Given this facility list concept, the algorithms have the ability to search for trunks, by facility, in an attempt to satisfy the bandwidth requirements of a given wideband call. If one facility does not have enough available bandwidth to support a given call, or it is not used for a given call due to the constraints presented in the following section, then the algorithm searches the next facility in the trunk group for the required bandwidth (if there is more than one facility in the trunk group).

In addition to searching for channels based on facilities and required bandwidth, Port Network (PN) preferential trunk routing is also employed. This PN routing applies within each algorithm at a higher priority than the constraints put on the algorithm by the parameters listed later in this section. In short, all facilities that reside on the same PN as the originating endpoint are searched in an attempt to satisfy the bandwidth of a given call, prior to searching any facilities on another PN.

Direction of trunk/hunting within facilities

The algorithms have the ability to select trunks from low B-channel to high B-channel or from high B-channel to low B-channel with an ISDN facility. This is a per ISDN trunk group option, but infers the direction of search within all ISDN facilities (or portions of those facilities) administered within that trunk group. This is necessary so the selection of trunks are not prone to as much glare as they otherwise would be if trunks were chosen in the same direction by both user and network sides of the ISDN interface. Note that in previous Avaya MultiVantage releases, the order in which trunks were selected, whether through linear or circular hunting, would always be with respect to the order in which trunks were administered within the trunk group. Now, with the support of wideband services, all trunks within an ISDN trunk group optioned for wideband are ordered based on this new “direction of trunk/hunt with facilities” parameter, and without regard to the order in which trunks are administered within the trunk group. If an ISDN trunk group is not optioned for wideband, then a cyclical trunk hunt based on the administration of trunks within the trunk group is still available.

H11

When a trunk group is administered to support H11, the algorithm to satisfy a call requiring 1,536 Kbps of bandwidth uses a fixed allocation scheme. That is, the algorithm searches for an available facility using the following facility-specific channel definitions.

- T1: H11 can only be carried on a facility without a D-channel being signaled in an NFAS arrangement (B-channels 1-24 are used).

- E1: Although the 1,536-kbps bandwidth could be satisfied using a number of fixed starting points (for example, 1, 2, 3, etc.) the only fixed starting point being supported is 1. Hence, B-channels 1–15 and 17–25 are always used to carry an H11 call on an E1 facility.

If the algorithm cannot find an available facility within the trunk group that meets these constraints, then the call is blocked from using this trunk group. In this case, the call may be routed to a different trunk group preference via Generalized Route Selection (GRS), at which time, based on the wideband options administered on that trunk group, the call would be subject to another hunt algorithm (that is, either the same H11 algorithm or perhaps an N x DS0 algorithm described in a later paragraph).

This same hunt algorithm, when offered any other call (other than a 1,920-kbps call) attempts to preserve idle facilities by selecting trunk(s) in a partially contaminated facility if one exists. If the bandwidth required by this call cannot be satisfied by any partially contaminated facility, then the call is placed on available trunk(s) within an idle facility, thus contaminating the facility. Again, facilities are selected via the trunk group's facility list and with PN preference, and trunk(s) within a facility are selected based on the direction of channel search administered. Note that on a T1 facility, a D-channel is not considered a busy trunk and results in a facility with a D-channel always being partially contaminated. On an E1 facility, however, a D-channel is not considered a busy trunk because H11 and H12 calls may still be placed on that facility; an E1 facility with a D-channel and idle B-channels is considered an idle facility.

H12

Since H12 is 1,920 Kbps which is comprised of 30 B-channels, a 1,920-kbps call can only be carried on an E1 facility. As with H11, the hunt algorithm uses a fixed allocation scheme with channel 1 being the fixed starting point. Hence, an H12 call always is carried on B-channels 1 to 15 and 17 to 31 on an E1 facility (as illustrated in the following table). When offered any other call (other than a 1,536-kbps call), the algorithm behaves as it does when H11 is optioned.

		DS0s Comprising Each Channel	
Facility	**ISDN Interface**	**H11**	**H12**
T1	23B + D	-	-
T1	24B (NFAS)	1-24	-
E1	30B + D	1-15, 17-25	1-15, 17-31
E1	31B (NFAS)	1-15, 17-25	1-15, 17-31

H0

When a trunk group is administered to support H0, the algorithm to satisfy a call requiring 384 Kbps of bandwidth also uses a fixed allocation scheme. Unlike the H11 fixed scheme which only supports a single fixed starting point, the H0 fixed scheme supports four (T1) or five (E1) fixed starting points. The H0 algorithm searches for an available quadrant within a facility based on the direction of trunk or hunt administered. If the algorithm cannot find an available quadrant within any facility allocated to this trunk group, then the call is blocked from using this trunk group. Again, based on GRS administration, the call may route to a different trunk group preference and be subject to another algorithm based on the wideband options administered.

This same trunk or hunt algorithm, when offered any narrowband or N x DS0 call, attempts to preserve idle quadrants by choosing a trunk(s) in a partially contaminated quadrant if one exists. If a partially contaminated quadrant capable of carrying the call does not exist, then the call is placed on available trunk(s) within an idle quadrant, thus contaminating the quadrant. Again, facilities are selected via the trunk group's facility list and with PN preference, and a trunk(s) within a facility is selected based on the direction administered. Note that a D-channel is considered a busy trunk and results in the top most quadrant of a T1, B-channels 19 to 24, always being partially contaminated. This is *not true* for NFAS.

If this H0 optioned trunk group is also administered to support H11, H12, or N x DS0, then this algorithm also attempts to preserve idle facilities. In other words, when offered a narrowband, H0, or N x DS0 call, the algorithm searches partially-contaminated facilities before it searches to idle facilities.

N x DS0

For the N x DS0 multi-rate service, a trunk group parameter determines whether a floating or a flexible trunk allocation scheme is to be used. The algorithm to satisfy an N x DS0 call is either floating or flexible.

- Floating (Contiguous) — In the floating scheme, an N x DS0 call is placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without any constraint being put on the starting channel (that is, no fixed starting point trunk).
- Flexible — In the flexible scheme, an N x DS0 call is placed on any set of B-channels as long as the requested bandwidth is satisfied. There is absolutely no constraint such as contiguity of B-channels or fixed starting points. Of course, as with all wideband calls, all the B-channels comprising the wideband call must reside on the same ISDN facility.

  Regardless of the allocation scheme employed, the N x DS0 algorithm, like the H11 and H12 algorithms, attempts to preserve idle facilities when offered B, H0, and N x DS0 calls. This is important so that N x DS0 calls, for large values of N, have a better chance of being satisfied by a given trunk group. However, if one of these calls cannot be satisfied by a partially-contaminated facility and an idle facility exists, a trunk on that idle facility is selected, thus contaminating that facility.

There are additional factors to note regarding specific values of N and the N x DS0 service:

- N = 1 — this is considered a narrowband call and is treated as any other voice or narrowband-data (B-channel) call.
- N = 6 — if a trunk group is optioned for both H0 and N x DS0 service, a 384-kbps call offered to that trunk group is treated as an H0 call and the H0 constraints apply. If the H0 constraints cannot be met, then the call is blocked.
- N = 24 — if a trunk group is optioned for both H11 and N x DS0 service, a 384-kbps call offered to that trunk group is treated as an H0 call and the H0 constraints apply. If the H0 constraints cannot be met, then the call is blocked.
- N = 24 — if a trunk group is optioned for both H11 and N x DS0 service, a 1,536-kbps call offered to that trunk group is treated as an H11 call and the H11 trunk allocation constraints apply.
- N = 30 — if a trunk group is optioned for both H12 and N x DS0 service, a 1,920-kbps call offered to that trunk group is treated as an H12 call and the H12 trunk allocation constraints apply.

Glare prevention

Glare occurs when both sides of an ISDN interface select the same B-channel for call initiation. For example, a user side of an interface selects the B-channel for an outgoing call and, before the switch receives and processes the SETUP message, the switch selects the same B-channel for call origination. Since wideband calls use more channels, the chances of glare are greater. Glare conditions can be limited with proper channel administration, but they may never be eliminated and some calls may still be dropped.

Some glare situations might not be resolvable. In one case, the network and the user side may send SETUP messages simultaneously or nearly simultaneously. Another glare scenario can occur in the brief window after the SETUP message has been sent but before the first response is received from the switch at the other side of the interface. If an incoming SETUP arrive during this window, the incoming SETUP message is allowed to proceed and the outgoing call is dropped. Various glare situations and their resolution are described in the following table.

Avaya MultiVantage does not negotiate channels for wideband calls.

GLARE RESOLUTION

Outgoing Call Type	Incoming Call Type	Switch-Supporting User Protocol	Switch-Supporting Network Protocol
B-channel	B-channel	No negotiation Incoming call (from network) wins	Negotiation is attempted Incoming call (from user) dropped if negotiation is unsuccessful
		Outgoing call (to network) retried on another trunk	Outgoing call (to user) stays up
B-channel(s)	Wide	No negotiation	No negotiation
		Incoming call (from network) dropped	Incoming call (from user) dropped
		Outgoing calls (to network) stay up but likely are dropped by network because channels are in use, although there is a possibility some switches might negotiate these calls.	Outgoing calls (to user) stay up and possibly stay up if other side lets the network call win.
Wide	B-channel(s)	No negotiation	Negotiation is attempted
		Incoming call (from network) wins	Incoming call (from user) dropped if negotiation is unsuccessful
		Outgoing call (to network) retried on another trunk	Outgoing call (to user) stays up
Wide	Wide	No negotiation	No negotiation
		Incoming call (from network) dropped	Incoming call (from user) dropped
		Outgoing call (to network) stays up but likely are dropped by network because channels are in use.	Outgoing call (to user) stays up and may not be dropped by other side because other side, if it is not an Avaya MultiVantage, may let the network call win.

To reduce glare probability, the network needs to be administered so both sides of the interface select channels from opposite ends of facilities. For example, on a

23B+D trunk group, the user side could be administered to select B-channels starting at channel 23 while the network side would be administered to start selecting at channel 1. Using the same example, if channel 22 is active but channel 23 is idle, the user side should select channel 23 for re-use. This is known as linear trunk hunt and is the hunt option used by Avaya MultiVantage for wideband.

Blocking prevention

Blocking occurs when insufficient B-channels required to make a call are available. Narrowband calls require only one channel so blocking is less likely than with wideband calls that require multiple B-channels. Blocking also occurs for wideband calls when bandwidth is not available in the appropriate format (that is, fixed, floating, or flexible).

To reduce blocking, the switch selects trunks for both wideband and narrowband calls to maximize availability of idle fixed channels for H0, H11, and H12 calls and idle floating channels for N x DS0 calls that require a contiguous bandwidth. The strategy for preserving idle channels depends on the channel type. The chances for blocking are reduced if you use a flexible algorithm, assuming it is supported on the other end.

Channel Type	Blocking Minimization Strategy
H0	Preserve idle quadrants
H11	Preserve idle facilities
H12	Preserve idle facilities
Flexible NxDS0	Preserve idle facilities
Floating NxDS0	Preserve idle facilities as first priority

Considerations

- For example, if the user side is provisioned to start at the high side (DS0 23) and DS0 22 is idle but DS0 23 is active, reselect DS0 22 for the next call. This is known as linear trunk hunting. Only the direction of hunt is administrable.

Interactions

- Administered Connections

  Provides call initiation for WAEs. All Administered Connections that originate from WAEs use the entire bandwidth administered for WAE. The destination of an Administered Connection can be a PRI endpoint.

- Automatic Circuit Assurance

  Treats wideband calls as single-trunk calls so that a single ACA-referral call is made if an ACA-referral call is required. The call is on the lowest B-channel associated with the wideband call.

- Call Coverage

  A wideband endpoint extension cannot be administered as a coverage point in a call-coverage path.

- Call Detail Recording

  When CDR is active for the trunk group, all wideband calls generate CDR records. The feature flag indicates a data call and CDR records contain bandwidth and Bearer Capability Class (BCC).

- Call Forwarding

  You must block Call Forwarding through Class of Service.

- Call Management System and Basic Call Management System

  Wideband calls can be carried over trunks that are measured by CMS and BCMS. Wideband endpoints are not measured by CMS and BCMS.

- Call Vectoring

  PRI endpoints use a vector-directory number when dialing. For example, PRI endpoint 1001 dials VDN 500. VDN 500 points to Vector 1. Vector 1 can point to other PRI endpoints such as route-to 1002, or route-to 1003, or busy.

  Call Vectoring is used by certain applications. When an incoming wideband call hunts for an available wideband endpoint, the call can route to a VDN, that sends the call to the first available PRI endpoint.

- Class of Restriction

  COR identifies caller and called-party privileges for PRI endpoints. Administer the COR so that account codes are not required. Forced entry of account codes is turned off for wideband endpoints.

- Class of Service

  COS determines the class of features that a wideband endpoint can activate.

- Facility and Non-Facility Associated Signaling

  FAS and NFAS with or without D-Channel Backup requires administration via signaling groups for trunk-side wideband interfaces.

- Facility Busy Indication

  You can administer a busy-indicator button for a wideband-endpoint extension, but the button does not accurately track endpoint status.

- Facility Test Calls

  Use Facility Test Calls to perform loop-back testing of the wideband call facility.

- Generalized Route Selection

  GRS supports wideband BCC to identify wideband calls. GRS searches a route pattern for a preference that has wideband BCC. Route preferences that support wideband BCC also support other BCCs to allow different call types to share the same trunk group.

- CO Trunk (TTC - Japan) Circuit Pack

  The CO Trunk (TTC - Japan) circuit pack cannot perform wideband switching. No member of the circuit pack should be a member of a wideband group.

World-Class Tone Detection and Generation

World-Class Tone Detection allows Avaya MultiVantage to identify and handle different types of call-progress tones. The tone detector and identification is used to display on Data Terminal Dialing and for deciding when to send digits on trunk calls through Abbreviated Dialing, ARS, AAR, and Data Terminal Dialing.

World-Class Tone Generation allows you to define call-progress tones. You can select values for frequency and cadence. If you do not define a call-progress tone, Avaya MultiVantage sends silence. Brief call-waiting tones are optimal because, while a tone is sounding, speech cannot be heard.

An Avaya representative must administer tone detection and tone generation when establishing country-specific system parameters.

Interactions

- Data Modules

  Multiline Data Terminal Dialing is disabled if the Multiple-line Level of Tone Detection field is **medium** or **broadband**. It is enabled if the Level of Tone Detection field is **precise**.

Glossary and abbreviations

Numerics

800 service
A service in the United States that allows incoming calls from certain areas to an assigned number for a flat-rate charge based on usage.

A

AA
Archangel. See angel.

AAC
ATM access concentrator

AAR
See Automatic Alternate Routing (AAR).

abandoned call
An incoming call in which the caller hangs up before the call is answered.

Abbreviated Dialing (AD)
A feature that allows callers to place calls by dialing just one or two digits.

AC
1. Alternating current.
2. See Administered Connection (AC).

AAR
Automatic Alternate Routing

ACA
See Automatic Circuit Assurance (ACA).

ACB
See Automatic Callback (ACB).

ACD
See Automatic Call Distribution (ACD).

ACD agent
See agent.

ACU
See Automatic calling unit (ACU)

ACW
See after-call work (ACW) mode.

access code
A 1-, 2-, or 3-digit dial code used to activate or cancel a feature, or access an outgoing trunk.

access endpoint
Either a nonsignaling channel on a DS1 interface or a nonsignaling port on an analog tie-trunk circuit pack that is assigned a unique extension.

access tie trunk
A trunk that connects a main communications system with a tandem communications system in an electronic tandem network (ETN). An access tie trunk can also be used to connect a system or tandem to a serving office or service node. Also called access trunk.

access trunk
See access tie trunk.

ACCUNET
A trademarked name for a family of digital services offered by AT&T in the United States.

ACD
See Automatic Call Distribution (ACD). ACD also refers to a work state in which an agent is on an ACD call.

ACD work mode
See work mode.

active-notification association
A link that is initiated by an adjunct, allowing it to receive event reports for a specific switch entity, such as an outgoing call.

active-notification call
A call for which event reports are sent over an active-notification association (communication channel) to the adjunct. Sometimes referred to as a monitored call.

active notification domain
VDN or ACD split extension for which event notification has been requested.

ACU
See Automatic calling unit (ACU).

AD
See Abbreviated Dialing (AD).

ADAP
AUDIX Data Acquisition Package

ADC
See analog-to-digital converter (ADC).

adjunct
A processor that does one or more tasks for another processor and that is optional in the configuration of the other processor. See also application.

adjunct-control association
A relationship initiated by an application via *Third Party Make Call*, the *Third Party Take Control,* or *Domain (Station) Control* capabilities to set up calls and control calls already in progress.

adjunct-controlled call
Call that can be controlled using an adjunct-control association. Call must have been originated via *Third Party Make Call* or *Domain (Station) Control* capabilities or must have been taken control of via *Third Party Take Control* or *Domain (Station) Control* capabilities.

adjunct-controlled split

An ACD split that is administered to be under adjunct control. Agents logged into such splits must do all telephony work, ACD login/ logout, and changes of work mode through the adjunct (except for auto-available adjunct-controlled splits, whose agents may not log in/out or change work mode).

adjunct-monitored call

An adjunct-controlled call, active-notification call, or call that provides event reporting over a domain-control association.

Adjunct-Switch Application Interface (ASAI)

A recommendation for interfacing adjuncts and communications systems, based on the CCITT Q.932 specification for layer 3.

ADM

Asynchronous data module

administer

To access and change parameters associated with the services or features of a system.

Administered Connection (AC)

A feature that allows the switch to automatically establish and maintain end-to-end connections between access endpoints (trunks) and/or data endpoints (data modules).

administration group

See capability group.

administration terminal

A terminal that is used to administer and maintain a system. See also terminal.

Administration Without Hardware (AWOH)

A feature that allows administration of ports without associated terminals or other hardware.

ADU

See asynchronous data unit (ADU).

AE

See access endpoint.

after-call work (ACW) mode

A mode in which agents are unavailable to receive ACD calls. Agents enter the ACW mode to perform ACD-related activities such as filling out a form after an ACD call.

AG

ASAI Gateway

agent

A person who receives calls directed to a split. A member of an ACD hunt group or ACD split. Also called an ACD agent.

agent report

A report that provides historical traffic information for internally measured agents.

AIM

Asynchronous interface module

AIOD

Automatic Identification of Outward Dialing

ALBO

Automatic Line Build Out

All trunks busy (ATB)
The state in which no trunks are available for call handling.

ALM-ACK
Alarm acknowledge

American Standard Code for Information Interchange
See ASCII (American Standard Code for Information Interchange).

AMW
Automatic Message Waiting

AN
Analog

analog
The representation of information by continuously variable physical quantities such as amplitude, frequency, and phase. See also digital.

analog data
Data that is transmitted over a digital facility in analog (PCM) form. The data must pass through a modem either at both ends or at a modem pool at the distant end.

analog telephone
A telephone that receives acoustic voice signals and sends analog electrical signals along the telephone line. Analog telephones are usually served by a single wire pair (tip and ring). The model-2500 telephone set is a typical example of an analog telephone.

analog-to-digital converter (ADC)
A device that converts an analog signal to digital form. See also digital-to-analog converter (DAC).

angel
A microprocessor located on each port card in a processor port network (PPN). The angel uses the control-channel message set (CCMS) to manage communications between the port card and the archangel on the controlling switch-processing element (SPE). The angel also monitors the status of other microprocessors on a port card and maintains error counters and thresholds.

ANI
See Automatic Number Identification (ANI).

ANSI
American National Standards Institute**.** A United States professional/technical association supporting a variety of standards.

answerback code
A number used to respond to a page from a code-calling or loudspeaker-paging system, or to retrieve a parked call.

AOL
Attendant-offered load

AP
Applications processor

APLT
Advanced Private-Line Termination

appearance

A software process that is associated with an extension and whose purpose is to supervise a call. An extension can have multiple appearances. Also called call appearance, line appearance, and occurrence. See also call appearance.

application

An adjunct that requests and receives ASAI services or capabilities. One or more applications can reside on a single adjunct. However, the switch cannot distinguish among several applications residing on the same adjunct and treats the adjunct, and all resident applications, as a single application. The terms application and adjunct are used interchangeably throughout this document.

applications processor

A micro-computer based, program controlled computer providing application services for the DEFINITY switch. The processor is used with several user-controlled applications such as traffic analysis and electronic documentation.

application service element

See capability group.

architecture

The organizational structure of a system, including hardware and software.

ARS

See Automatic Route Selection (ARS).

ASAI

See Adjunct-Switch Application Interface (ASAI)

ASCII (American Standard Code for Information Interchange)

The standard code for representing characters in digital form. Each character is represented by an 8-bit code (including parity bit).

association

A communication channel between adjunct and switch for messaging purposes. An active association is one that applies to an existing call on the switch or to an extension on the call.

asynchronous data transmission

A method of transmitting data in which each character is preceded by a start bit and followed by a stop bit, thus permitting data characters to be transmitted at irregular intervals. This type transmission is advantageous when transmission is not regular (characters typed at a keyboard). Also called asynchronous transmission. See also synchronous data transmission.

asynchronous data unit (ADU)

A device that allows direct connection between RS-232C equipment and a digital switch.

Asynchronous Transfer Mode (ATM)

A packet-like switching technology in which data is transmitted in fixed-size (53-byte) cells. ATM provides high-speed access for data communication in LAN, campus, and WAN environments.

ATB

See All trunks busy (ATB).

ATD

See Attention dial (ATD).

attendant

A person at a console who provides personalized service for incoming callers and voice-services users by performing switching and signaling operations. See also attendant console.

ATM
See Asynchronous Transfer Mode (ATM).

attendant console
The workstation used by an attendant. The attendant console allows the attendant to originate a call, answer an incoming call, transfer a call to another extension or trunk, put a call on hold, and remove a call from hold. Attendants using the console can also manage and monitor some system operations. Also called console. See also attendant.

Attention dial (ATD)
A command in the Hayes modem command set for asynchronous modems.

Audio Information Exchange (AUDIX)
A fully integrated voice-mail system. Can be used with a variety of communications systems to provide call-history data, such as subscriber identification and reason for redirection.

AUDIX
See Audio Information Exchange (AUDIX).

auto-in trunk group
Trunk group for which the CO processes all of the digits for an incoming call. When a CO seizes a trunk from an auto-in trunk group, the switch automatically connects the trunk to the destination — typically an ACD split where, if no agents are available, the call goes into a queue in which callers are answered in the order in which they arrive.

Auto-In Work mode
One of four agent work modes: the mode in which an agent is ready to process another call as soon as the current call is completed.

Automatic Alternate Routing (AAR)
A feature that routes calls to other than the first-choice route when facilities are unavailable.

Automatic Callback (ACB)
A feature that enables internal callers, upon reaching a busy extension, to have the system automatically connect and ring both parties when the called party becomes available.

Automatic Call Distribution (ACD)
A feature that answers calls, and then, depending on administered instructions, delivers messages appropriate for the caller and routes the call to an agent when one becomes available.

Automatic Call Distribution (ACD) split
A method of routing calls of a similar type among agents in a call center. Also, a group of extensions that are staffed by agents trained to handle a certain type of incoming call.

Automatic calling unit (ACU)
A device that places a telephone call.

Automatic Circuit Assurance (ACA)
A feature that tracks calls of unusual duration to facilitate troubleshooting. A high number of very short calls or a low number of very long calls may signify a faulty trunk.

Automatic Number Identification (ANI)
Representation of the calling number, for display or for further use to access information about the caller.

automatic restoration
A service that restores disrupted connections between access endpoints (nonsignaling trunks) and data endpoints (devices that connect the switch to data terminal and/or communications equipment). Restoration is done within seconds of a service disruption so that critical data applications can remain operational.

Automatic Route Selection (ARS)
A feature that allows the system to automatically choose the least-cost way to send a toll call.

automatic trunk
A trunk that does not require addressing information because the destination is predetermined. A request for service on the trunk, called a seizure, is sufficient to route the call. The normal destination of an automatic trunk is the communications-system attendant group. Also called automatic incoming trunk and automatic tie trunk.

AUX
Auxiliary

auxiliary equipment
Equipment used for optional system features, such as Loudspeaker Paging and Music-on-Hold.

auxiliary trunk
A trunk used to connect auxiliary equipment, such as radio-paging equipment, to a communications system.

Aux-Work mode
A work mode in which agents are unavailable to receive ACD calls. Agents enter Aux-Work mode when involved in non-ACD activities such as taking a break, going to lunch, or placing an outgoing call.

AVD
Alternate voice/data

AWOH
See Administration Without Hardware (AWOH).

AWG
American Wire Gauge

AWT
Average work time

B

B8ZS
Bipolar Eight Zero Substitution.

bandwidth
The difference, expressed in hertz, between the defined highest and lowest frequencies in a range.

barrier code
A security code used with the Remote Access feature to prevent unauthorized access to the system.

Basic Rate Interface (BRI)
A standard ISDN frame format that specifies the protocol used between two or more communications systems. BRI runs at 192 Mbps and provides two 64-kbps B-channels (voice and data) and one 16-kbps D-channel (signaling). The D-channel connects, monitors, and disconnects all calls. It also can carry low-speed packet data at 9.6 kbps.

baud
A unit of transmission rate equal to the number of signal events per second. See also bit rate and bits per second (bps).

BCC
See Bearer capability class (BCC).

BCMS
Basic Call Management System

BCT
See business communications terminal (BCT).

Bearer capability class (BCC)
Code that identifies the type of a call (for example, voice and different types of data). Determination of BCC is based on the caller's characteristics for non-ISDN endpoints and on the Bearer Capability and Low-Layer Compatibility Information Elements of an ISDN endpoint. Current BCCs are 0 (voice-grade data and voice), 1 (DMI mode 1, 56 kbps data transmission), 2 (DMI mode 2, synchronous/asynchronous data transmission up to 19.2 kbps) 3 (DMI mode 3, 64 kbps circuit/packet data transmission), 4 (DMI mode 0, 64 kbps synchronous data), 5 (temporary signaling connection, and 6 (wideband call, 128–1984 kbps synchronous data).

BER
Bit error rate

BHCC
Busy-hour call completions

bit (binary digit)
One unit of information in binary notation, having two possible values: 0 or 1.

bits per second (bps)
The number of binary units of information that are transmitted or received per second. See also baud and **bit rate**.

bit rate
The speed at which bits are transmitted, usually expressed in bits per second. Also called data rate. See also baud and bits per second (bps).

BLF
Busy Lamp Field

BN
Billing number

BOS
Bit-oriented signaling

BPN
Billed-party number

bps
See bits per second (bps).

bridge (bridging)
The appearance of a voice terminal's extension at one or more other voice terminals.

BRI
The ISDN Basic Rate Interface specification.

bridged appearance
A call appearance on a voice terminal that matches a call appearance on another voice terminal for the duration of a call.

BTU

British Thermal Unit

buffer

1. In hardware, a circuit or component that isolates one electrical circuit from another. Typically, a buffer holds data from one circuit or process until another circuit or process is ready to accept the data.
2. In software, an area of memory that is used for temporary storage.

bus

A multiconductor electrical path used to transfer information over a common connection from any of several sources to any of several destinations.

business communications terminal (BCT)

A digital data terminal used for business applications. A BCT can function via a data module as a special-purpose terminal for services provided by a processor or as a terminal for data entry and retrieval.

BX.25

A version of the CCITT X.25 protocol for data communications. BX.25 adds a fourth level to the standard X.25 interface. This uppermost level combines levels 4, 5, and 6 of the ISO reference model.

bypass tie trunks

A 1-way, outgoing tie trunk from a tandem switch to a main switch in an ETN. Bypass tie trunks, provided in limited quantities, are used as a last-choice route when all trunks to another tandem switch are busy. Bypass tie trunks are used only if all applicable intertandem trunks are busy.

byte

A sequence of (usually eight) bits processed together.

C

cabinet

Housing for racks, shelves, or carriers that hold electronic equipment.

cable

Physical connection between two pieces of equipment (for example, data terminal and modem) or between a piece of equipment and a termination field.

cable connector

A jack (female) or plug (male) on the end of a cable. A cable connector connects wires on a cable to specific leads on telephone or data equipment.

CACR

Cancellation of Authorization Code Request

CAG

Coverage answer group

call appearance

1. For the attendant console, six buttons, labeled a–f, used to originate, receive, and hold calls. Two lights next to the button show the status of the call appearance.
2. For the voice terminal, a button labeled with an extension and used to place outgoing calls, receive incoming calls, or hold calls. Two lights next to the button show the status of the call appearance.

call-control capabilities

Capabilities (*Third Party Selective Hold*, *Third Party Reconnect*, *Third Party Merge*) that can be used in either of the Third Party Call Control ASE (cluster) subsets (Call Control and Domain Control).

Call Detail Recording (CDR)
A feature that uses software and hardware to record call data (same as CDRU).

Call Detail Recording utility (CDRU)
Software that collects, stores, optionally filters, and outputs call-detail records.

Call Management System (CMS)
An application, running on an adjunct processor, that collects information from an ACD unit. CMS enables customers to monitor and manage telemarketing centers by generating reports on the status of agents, splits, trunks, trunk groups, vectors, and VDNs, and enables customers to partially administer the ACD feature for a communications system.

call-reference value (CRV)
An identifier present in ISDN messages that associates a related sequence of messages. In ASAI, CRVs distinguish between associations.

call vector
A set of up to 15 vector commands to be performed for an incoming or internal call.

callback call
A call that automatically returns to a voice-terminal user who activated the Automatic Callback or Ringback Queuing feature.

call-waiting ringback tone
A low-pitched tone identical to ringback tone except that the tone decreases in the last 0.2 seconds (in the United States). Call-waiting ringback tone notifies the attendant that the Attendant Call Waiting feature is activated and that the called party is aware of the waiting call. Tones in international countries may sound different.

call work code
A number, up to 16 digits, entered by ACD agents to record the occurrence of customer-defined events (such as account codes, social security numbers, or phone numbers) on ACD calls.

CAMA
Centralized Automatic Message Accounting

carrier
An enclosed shelf containing vertical slots that hold circuit packs.

carried load
The amount of traffic served by traffic-sensitive facilities during a given interval.

CARR-POW
Carrier Port and Power Unit for AC Powered Systems

CAS
Centralized Attendant Service or Call Accounting System

capability
A request or indication of an operation. For example, *Third Party Make Call* is a request for setting up a call; *event report* is an indication that an event has occurred.

capability group

Set of capabilities, determined by switch administration, that can be requested by an application. Capability groups denote association types. For example, *Call Control* is a type of association that allows certain functions (the ones in the capability group) to be performed over this type of association. Also referred to as administration groups or application service elements (ASEs).

CA-TSC

Call-Associated Temporary Signaling Connection

cause value

A value is returned in response to requests or in event reports when a denial or unexpected condition occurs. ASAI cause values fall into two coding standards: Coding Standard 0 includes any cause values that are part of AT&T and CCITT ISDN specifications; Coding standard 3 includes any other ASAI cause values. This document uses a notation for cause value where the coding standard for the cause is given first, then a slash, then the cause value. Example: CS0/100 is coding standard 0, cause value 100.

CBC

Call-by-call or coupled bonding conductor

CC

Country code

CCIS

Common-Channel Interoffice Signaling

CCITT

CCITT (Comitte Consultatif International Telephonique et Telegraphique), now called *International Telecommunications Union* (ITU). See International Telecommunications Union (ITU).

CCMS

Control-Channel Message Set

CCS

See capability.

CCS or hundred call seconds

A unit of call traffic. Call traffic for a facility is scanned every 100 seconds. If the facility is busy, it is assumed to have been busy for the entire scan interval. There are 3600 seconds per hour. The Roman numeral for 100 is the capital letter C. The abbreviation for call seconds is CS. Therefore, 100 call seconds is abbreviated CCS. If a facility is busy for an entire hour, then it is said to have been busy for 36 CCS. See also **Erlang.**

CCSA

Common-Control Switching Arrangement

CDM

Channel-division multiplexing

CDOS

Customer-dialed and operator serviced

CDPD

Customer database-provided digits

CDR

See Call Detail Recording (CDR).

CDRP

Call Detail Record Poller

CDRR
Call Detail Recording and Reporting

CDRU
See Call Detail Recording utility (CDRU).

CED
Caller entered digits

CEM
Channel-expansion multiplexing

center-stage switch (CSS)
The central interface between the processor port network and expansion port networks in a CSS-connected system.

central office (CO)
The location housing telephone switching equipment that provides local telephone service and access to toll facilities for long-distance calling.

central office (CO) codes
The first three digits of a 7-digit public-network telephone number in the United States.

central office (CO) trunk
A telecommunications channel that provides access from the system to the public network through the local CO.

CEPT1
European Conference of Postal and Telecommunications Rate 1

CESID
Caller's Emergency Service Identification

channel
1. A circuit-switched call.
2. A communications path for transmitting voice and data.
3. In wideband, all of the time slots (contiguous or noncontiguous) necessary to support a call. Example: an H0-channel uses six 64-kbps time slots.
4. A DS0 on a T1 or E1 facility not specifically associated with a logical circuit-switched call; analogous to a single trunk.

channel negotiation
The process by which the channel offered in the Channel Identification Information Element (CIIE) in the SETUP message is negotiated to be another channel acceptable to the switch that receives the SETUP message and ultimately to the switch that sent the SETUP. Negotiation is attempted only if the CIIE is encoded as *Preferred*. Channel negotiation is not attempted for wideband calls.

CI
Clock input

circuit
1. An arrangement of electrical elements through which electric current flows.
2. A channel or transmission path between two or more points.

circuit pack
A card on which electrical circuits are printed, and IC chips and electrical components are installed. A circuit pack is installed in a switch carrier.

CISPR

International Special Committee on Radio Interference

Class of Restriction (COR)

A feature that allows up to 96 classes of call-origination and call-termination restrictions for voice terminals, voice-terminal groups, data modules, and trunk groups. See also Class of Service (COS).

Class of Service (COS)

A feature that uses a number to specify if voice-terminal users can activate the Automatic Callback, Call Forwarding All Calls, Data Privacy, or Priority Calling features. See also Class of Restriction (COR).

cm

Centimeter

CM

Connection Manager

CMC

Compact Modular Cabinet

CMDR

Centralized Message Detail Recording

CMS

Call Management System

CO

See central office (CO).

common-control switching arrangement (CCSA)

A private telecommunications network using dedicated trunks and a shared switching center for interconnecting company locations.

communications system

The software-controlled processor complex that interprets dialing pulses, tones, and keyboard characters and makes the proper connections both within the system and external to the system. The communications system itself consists of a digital computer, software, storage device, and carriers with special hardware to perform the connections. A communications system provides voice and data communications services, including access to public and private networks, for telephones and data terminals on a customer's premises. See also switch.

confirmation tone

A tone confirming that feature activation, deactivation, or cancellation has been accepted.

connectivity

The connection of disparate devices within a single system.

console

See attendant console.

contiguous

Adjacent DS0s within one T1 or E1 facility or adjacent TDM or fiber time slots. The first and last TDM bus, DS0, or fiber time slots are not considered contiguous (no wraparound). For an E1 facility with a D-channel, DS0s 15 and 17 are considered contiguous.

control cabinet

See control carrier.

control carrier
A carrier in a multi-carrier cabinet that contains the SPE circuit packs and, unlike an G3r control carrier, port circuit packs. Also called control cabinet in a single-carrier cabinet. See also switch-processing element (SPE).

controlled station
A station that is monitored and controlled via a domain-control association.

COR
See Class of Restriction (COR).

COS
See Class of Service (COS).

coverage answer group
A group of up to eight voice terminals that ring simultaneously when a call is redirected to it by Call Coverage. Any one of the group can answer the call.

coverage call
A call that is automatically redirected from the called party's extension to an alternate answering position when certain coverage criteria are met.

coverage path
The order in which calls are redirected to alternate answering positions.

coverage point
An extension or attendant group, VDN, or ACD split designated as an alternate answering position in a coverage path.

covering user
A person at a coverage point who answers a redirected call.

CP
Circuit pack

CPE
Customer-premises equipment

CPN
Called-party number

CPN/BN
Calling-party number/billing number

CPTR
Call-progress-tone receiver

CRC
Cyclical Redundancy Checking

critical-reliability system
A system that has the following duplicated items: control carriers, tone clocks, EI circuit packs, and cabling between port networks and center-stage switch in a CSS-connected system. See also duplicated common control, and duplication.

CSA
Canadian Safety Association or Customer Software Administrator

CSCC
Compact single-carrier cabinet

CSCN
Center-stage control network

CSD
Customer-service document

CSM
Centralized System Management

CSS
See center-stage switch (CSS).

CSSO
Customer Services Support Organization

CSU
Channel service unit

CTI Station
CTI Stations are AWOH stations that allow applications to originate and receive calls. CTI Stations support ASAI call control features such as hold, answer, drop, conference etc. Calls on a CTI station operate the same way as they would on a real phone. CTI Stations can also be used to originate phantom calls.

CTS
Clear to Send

CWC
See call work code.

D

DAC
1. Dial access code or Direct Agent Calling
2. See digital-to-analog converter (DAC).

data channel
A communications path between two points used to transmit digital signals.

data-communications equipment (DCE)
The equipment (usually a modem, data module, or packet assembler/disassembler) on the network side of a communications link that makes the binary serial data from the source or transmitter compatible with the communications channel.

data link
The configuration of physical facilities enabling end terminals to communicate directly with each other.

data module
An interconnection device between a BRI or DCP interface of the switch and data terminal equipment or data communications equipment.

data path
The end-to-end connection used for a data communications link. A data path is the combination of all elements of an interprocessor communication in a DCS.

data port
A point of access to a computer that uses trunks or lines for transmitting or receiving data.

data rate
See bit rate.

data service unit (DSU)
A device that transmits digital data on transmission facilities.

data terminal
An input/output (I/O) device that has either switched or direct access to a host computer or to a processor interface.

data terminal equipment (DTE)
Equipment consisting of the endpoints in a connection over a data circuit. In a connection between a data terminal and host, the terminal, the host, and their associated modems or data modules make up the DTE.

dB
Decibel

dBA
Decibels in reference to amperes.

dBrnC
Decibels above reference noise with C filter.

DC
Direct current

DCE
Data-communications equipment

D-channel backup
Type of backup used with Non-Facility Associated Signaling (NFAS). A primary D-channel provides signaling for an NFAS D-channel group (two or more PRI facilities). A second D-channel, on a separate PRI facility of the NFAS D-channel group, is designated as backup for the D-channel. Failure of the primary D-channel causes automatic transfer of call-control signaling to the backup D-channel. The backup becomes the primary D-channel. When the failed channel returns to service, it becomes the backup D-channel.

DCO
Digital central office

DCP
Digital Communications Protocol

DCS
Distributed Communications System

DDC
Direct Department Calling

DDD
Direct Distance Dialing

delay-dial trunk
A trunk that allows dialing directly into a communications system (digits are received as they are dialed).

denying a request
Sending a negative acknowledgment (NAK), done by sending an FIE with a *return error* component (and a cause value). It should not be confused with the denial event report that applies to calls.

designated voice terminal

The specific voice terminal to which calls, originally directed to a certain extension, are redirected. Commonly used to mean the forwarded-to terminal when Call Forwarding All Calls is active.

dial-repeating trunks

A PBX tie trunk that is capable of handling PBX station-signaling information without attendant assistance.

dial-repeating tie trunk

A tie trunk that transmits called-party addressing information between two communications systems.

DID

Direct Inward Dialing

digit conversion

A process used to convert specific dialed numbers into other dialed numbers.

digital

The representation of information by discrete steps. See also analog.

digital communications protocol (DCP)

A proprietary protocol used to transmit both digitized voice and digitized data over the same communications link. A DCP link is made up of two 64-kbps information (I-) channels and one 8-kbps signaling (S-) channel. The DCP protocol supports two information-bearing channels, and thus two telephones/data modules. The I1 channel is the DCP channel assigned on the first page of the 8411 station form. The I2 channel is the DCP channel assigned on the analog adjunct page of the 8411 station form or on the data module page.

Digital Communications Protocol. The DCP protocol supports two information-bearing channels, and thus two telephones/data modules. The I1 channel is the DCP channel assigned on the first page of the 8411 station form. The I2 channel is the DCP channel assigned on the analog adjunct page of the 8411 station form or on the data module page.

digital data endpoints

In DEFINITY ECS, devices such as the 510D terminal or the 515-type business communications terminal (BCT).

digital multiplexed interface (DMI)

An interface that provides connectivity between a communications system and a host computer or between two communications systems using DS1 24th-channel signaling. DMI provides 23 64-kbps data channels and 1 common-signaling channel over a twisted-pair connection. DMI is offered through two capabilities: bit-oriented signaling (DMI-BOS) and message-oriented signaling (DMI-MOS).

digital signal level 0 (DS0)

A single 64-kbps voice channel. A DS0 is a single 64-kbps channel in a T1 or E1 facility and consists of eight bits in a T1 or E1 frame every 125 microseconds.

digital signal level 1 (DS1)

A single 1.544-Mbps (United States) or 2.048-Mbps (outside the United States) digital signal carried on a T1 transmission facility. A DS1 converter complex consists of a pair, one at each end, of DS1 converter circuit packs and the associated T1/E1 facilities.

digital terminal data module (DTDM)

An integrated or adjunct data module that shares with a digital telephone the same physical port for connection to a communications system. The function of a DTDM is similar to that of a PDM and MPDM in that it converts RS-232C signals to DCP signals.

digital-to-analog converter (DAC)

A device that converts data in digital form to the corresponding analog signals. See also analog-to-digital converter (ADC).

digital transmission
A mode of transmission in which information to be transmitted is first converted to digital form and then transmitted as a serial stream of pulses.

digital trunk
A circuit that carries digital voice and/or digital data in a telecommunications channel.

DIOD
Direct Inward and Outward Dialing

direct agent
A feature, accessed only via ASAI, that allows a call to be placed in a split queue but routed only to a specific agent in that split. The call receives normal ACD call treatment (for example, announcements) and is measured as an ACD call while ensuring that a particular agent answers.

Direct Extension Selection (DXS)
A feature on an attendant console that allows an attendant direct access to voice terminals by pressing a group-select button and a DXS button.

Direct Inward Dialing (DID)
A feature that allows an incoming call from the public network (not FX or WATS) to reach a specific telephone without attendant assistance.

Direct Inward Dialing (DID) trunk
An incoming trunk used for dialing directly from the public network into a communications system without help from the attendant.

disk drive
An electromechanical device that stores data on and retrieves data from one or more disks.

distributed communications system (DCS)
A network configuration linking two or more communications systems in such a way that selected features appear to operate as if the network were one system.

DIVA
Data In/Voice Answer

DLC
Data line circuit

DLDM
Data-line data module

DMI
Digital-multiplexed interface

DND
Do not disturb

DNIS
Dialed-Number Identification Service

DOD
Direct Outward Dialing

domain
VDNs, ACD splits, and stations. The VDN domain is used for active-notification associations. The ACD-split domain is for active-notification associations and domain-control associations. The station domain is used for the domain-control associations.

domain-control association

A *Third Party Domain Control Request* capability initiates a unique CRV/link number combination, which is referred to as a domain-control association.

domain-controlled split

A split for which *Third Party Domain Control* request has been accepted. A domain-controlled split provides an event report for logout.

domain-controlled station

A station for which a *Third_Party_Domain_Control* request has been accepted. A domain-controlled station provides event reports for calls that are alerting, connected, or held at the station.

domain-controlled station on a call

A station that is active on a call, and which provides event reports over one or two domain-control associations.

DOSS

Delivery Operations Support System

DOT

Duplication Option Terminal

DPM

Dial Plan Manager

DPR

Dual-port RAM

DS1

Digital Signal Level 1

DS1C

Digital Signal Level-1 protocol C

DS1 CONV

Digital Signal Level-1 converter

DSI

Digital signal interface

DSU

Data service unit

DTDM

Digital-terminal data module

DTE

Data-terminal equipment

DTGS

Direct Trunk Group Select

DTMF

Dual-tone multifrequency

DTS

Disk-tape system

duplicated common control
Two processors ensuring continuous operation of a communications system. While one processor is online, the other functions as a backup. The backup processor goes online periodically or when a problem occurs.

duplication
The use of redundant components to improve availability. When a duplicated subsystem fails, its backup redundant system automatically takes over.

duplication option
A system option that duplicates the following: control carrier containing the SPE, EI circuit packs in carriers, fiber-optic cabling between port networks, and center-stage switch in a CSS-connected system.

DWBS
DEFINITY Wireless Business System

DXS
Direct extension selection

E

E1
A digital transmission standard that carries traffic at 2.048 Mbps. The E1 facility is divided into 32 channels (DS0s) of 64 kbps information. Channel 0 is reserved for framing and synchronization information. A D-channel occupies channel 16.

E & M
Ear and mouth (receive and transmit)

EA
Expansion archangel

EAL
Expansion archangel link

ear and mouth (E & M) signaling
Trunk supervisory signaling, used between two communications systems, whereby signaling information is transferred through 2-state voltage conditions (on the E and M leads) for analog applications and through a single bit for digital applications.

EAS
See "Expert Agent Selection".

ECC
Error Correct Code

ECMA
European Computer Manufacturers Association

EEBCDIC
Extended Binary-Coded Decimal Interexchange Code

EFP
Electronic power feed

EI
Expansion interface

EIA

Electronic Industries Association

EIA-232

A physical interface specified by the EIA. EIA-232 transmits and receives asynchronous data at speeds of up to 19.2 kbps over cable distances of up to 50 feet. EIA-232 replaces RS-232 protocol in some DEFINITY applications.

electronic tandem network (ETN)

A tandem tie-trunk network that has automatic call-routing capabilities based on the number dialed and the most preferred route available. Each switch in the network is assigned a unique private network office code (RNX), and each voice terminal is assigned a unique extension.

Electronics Industries Association (EIA)

A trade association of the electronics industry that establishes electrical and functional standards.

emergency transfer

If a major system failure occurs, automatic transfer is initiated to a group of telephones capable of making outgoing calls. The system operates in this mode until the failure is repaired and the system automatically returns to normal operation. Also called power-failure transfer.

EMI

Electromagnetic interference

end-to-end signaling

The transmission of touch-tone signals generated by dialing from a voice terminal to remote computer equipment. These digits are sent over the trunk as DTMF digits whether the trunk signaling type is marked as tone or rotary and whether the originating station is tone or rotary. Example: a call to a voice-mail machine or automated-attendant service. A connection is first established over an outgoing trunk. Then additional digits are dialed to transmit information to be processed by the computer equipment.

enhanced private-switched communications service (EPSCS)

An analog private telecommunications network based on the No. 5 crossbar and 1A ESS that provides advanced voice and data telecommunications services to companies with many locations.

EPN

Expansion-port network

EPROM

Erasable programmable read-only memory

EPSCS

Enhanced Private Switched Communications Services

ERL

Echo return loss

Erlang

A unit of traffic intensity, or load, used to express the amount of traffic needed to keep one facility busy for one hour. One Erlang is equal to 36 CCS. See also capability.

ESF

Extended superframe format

ESPA

European Standard Paging Access

ETA

Extended Trunk Access; also Enhanced Terminal Administration

ETN
Electronic tandem network

ETSI
European Telecommunications Standards Institute

expansion archangel (EAA)
A network-control microprocessor located on an expansion interface (EI) port circuit pack in an expansion port network. The EA provides an interface between the EPN and its controlling switch-processing element.

expansion-archangel link (EAL)
A link-access function on the D-channel (LAPD) logical link that exists between a switch-processing element and an expansion archangel (EA). The EAL carries control messages from the SPE to the EA and to port circuit packs in an expansion port network.

expansion control cabinet
See expansion control carrier.

expansion control carrier
A carrier in a multicarrier cabinet that contains extra port circuit packs and a maintenance interface. Also called expansion control cabinet in a single-carrier cabinet.

expansion interface (EI)
A port circuit pack in a port network that provides the interface between a PN's TDM bus/ packet bus and a fiber-optic link. The EI carries circuit-switched data, packet-switched data, network control, timing control, and DS1 control. In addition, an EI in an expansion port network communicates with the master maintenance circuit pack to provide the EPN's environmental and alarm status to the switch-processing element.

expansion port network (EPN)
A port network (PN) that is connected to the TDM bus and packet bus of a processor port network (PPN). Control is achieved by indirect connection of the EPN to the PPN via a port-network link (PNL). See also port network (PN).

Expert Agent Selection
A feature allowing incoming calls to be routed to specialized groups of agents within a larger pool of agents.

extension-in
Extension-In (ExtIn) is the work state agents go into when they answer (receive) a non-ACD call. If the agent is in Manual-In or Auto-In and receives an extension-in call, it is recorded by CMS as an AUX-In call.

extension-out
The work state that agents go into when they place (originate) a non-ACD call.

external measurements
Those ACD measurements that are made by the External CMS adjunct.

extension
A 1- to 5-digit number by which calls are routed through a communications system or, with a Uniform Dial Plan (UDP) or main-satellite dialing plan, through a private network.

external call
A connection between a communications system user and a party on the public network or on another communications system in a private network.

F

FAC
Feature Access Code

facility
A telecommunications transmission pathway and associated equipment.

facility-associated signaling (FAS)
Signaling for which a D-channel carries signaling only for those channels on the same physical interface.

FAS
Facility-associated signaling

FAT
Facility access trunk

FAX
Facsimile

FCC
Federal Communications Commission

FEAC
Forced Entry of Account Codes

feature
A specifically defined function or service provided by the system.

feature button
A labeled button on a telephone or attendant console used to access a specific feature.

FEP
Front-end processor

fiber optics
A technology using materials that transmit ultra wideband electromagnetic light-frequency ranges for high-capacity carrier systems.

FIC
Facility interface codes

fixed
A trunk allocation term. In the fixed allocation scheme, the time slots necessary to support a wideband call are contiguous, and the first time slot is constrained to certain starting points.

flexible
A trunk allocation term. In the flexible allocation scheme, the time slots of a wideband call can occupy non-contiguous positions within a single T1 or E1 facility.

floating
A trunk allocation term. In the floating allocation scheme, the time slots of a wideband call are contiguous, but the position of the first time slot is not fixed.

FNPA
Foreign Numbering-Plan Area

foreign-exchange (FX)
A CO other than the one providing local access to the public telephone network.

foreign-exchange trunk
A telecommunications channel that directly connects the system to a CO other than its local CO.

foreign numbering-plan area code (FNPAC)
An area code other than the local area code, that must be dialed to call outside the local geographical area.

FRL
Facilities Restriction Level

FX
Foreign exchange

G

G3-MA
Generic 3 Management Applications

G3-MT
Generic 3 Management Terminal

G3r
Generic 3, RISC (Reduced Instruction Set Computer)

generalized route selection (GRS)
An enhancement to Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) that performs routing based on call attributes, such as Bearer Capability Classes (BCCs), in addition to the address and facilities restriction level (FRL), thus facilitating a Uniform Dial Plan (UDP) that is independent of the type of call being placed.

glare
The simultaneous seizure of a 2-way trunk by two communications systems, resulting in a standoff.

GM
Group manager

GPTR
General-purpose tone receiver

grade of service
The number of call attempts that fail to receive service immediately. Grade of service is also expressed as the quantity of all calls that are blocked or delayed.

ground-start trunk
A trunk on which, for outgoing calls, the system transmits a request for services to a distant switching system by grounding the trunk ring lead. To receive the digits of the called number, that system grounds the trunk tip lead. When the system detects this ground, the digits are sent.

GRS
Generalized Route Selection

H

H0
An ISDN information transfer rate for 384-kbps data defined by CCITT and ANSI standards.

H11
An ISDN information transfer rate for 1536-kbps data defined by CCITT and ANSI standards.

H12
An ISDN information transfer rate for 1920-kbps data defined by CCITT and ANSI standards.

handshaking logic
A format used to initiate a data connection between two data module devices.

hertz (Hz)
A unit of frequency equal to one cycle per second.

high-reliability system
A system having the following: two control carriers, duplicate expansion interface (EI) circuit packs in the PPN (in G3r with CSS), and duplicate switch node clock circuit packs in the switch node (SN) carriers. See also duplicated common control, duplication, duplication option, and critical-reliability system.

HNPA
See home numbering-plan area code (HNPA).

holding time
The total length of time in minutes and seconds that a facility is used during a call.

home numbering-plan area code (HNPA)
The local area code. The area code does not have to be dialed to call numbers within the local geographical area.

hop
Nondirect communication between two switch communications interfaces (SCI) where the SCI message passes automatically without intermediate processing through one or more intermediate SCIs.

host computer
A computer, connected to a network, that processes data from data-entry devices.

hunt group
A group of extensions that are assigned the Station Hunting feature so that a call to a busy extension reroutes to an idle extension in the group. See also ACD work mode.

Hz
See hertz (Hz).

I

I1
The first information channel of DCP.

I2
The second information channel of DCP.

I2 Interface
A proprietary interface used for the DEFINITY Wireless Business System for the radio-controller circuit packs. Each interface provides communication between the radio-controller circuit pack and up to two wireless fixed bases.

I3 Interface
A proprietary interface used for the DEFINITY Wireless Business System for the cell antenna units. Each wireless fixed base can communicate to up to four cell antenna units.

IAS
Inter-PBX Attendant Service

ICC
Intercabinet cable or intercarrier cable

ICD
Inbound Call Director

ICDOS
International Customer-Dialed Operator Service

ICHT
Incoming call-handling table

ICI
Incoming call identifier

ICM
Inbound Call Management

IDDD
International Direct Distance Dialing

IDF
Intermediate distribution frame

IE
Information element

immediate-start tie trunk
A trunk on which, after making a connection with a distant switching system for an outgoing call, the system waits a nominal 65 ms before sending the digits of the called number. This allows time for the distant system to prepare to receive digits. On an incoming call, the system has less than 65 ms to prepare to receive the digits.

IMT
Intermachine trunk

in
Inch

INADS
Initialization and Administration System

ICLID
Incoming Caller ID

incoming gateway
A PBX that routes an incoming call on a trunk *not* administered for Supplementary Services Protocol B to a trunk *not* administered for Supplementary Services Protocol B.

information exchange
The exchange of data between users of two different systems, such as the switch and a host computer, over a LAN.

Information Systems Network (ISN)
A WAN and LAN with an open architecture combining host computers, minicomputers, word processors, storage devices, PCs, high-speed printers, and nonintelligent terminals into a single packet-switching system.

INS
ISDN Network Service

inside call
A call placed from one telephone to another within the local communications system.

Integrated Services Digital Network (ISDN)
A public or private network that provides end-to-end digital communications for all services to which users have access by a limited set of standard multipurpose user-network interfaces defined by the CCITT. Through internationally accepted standard interfaces, ISDN provides digital circuit-switched or packet-switched communications within the network and links to other ISDNs to provide national and international digital communications. See also Integrated Services Digital Network Basic Rate Interface (ISDN-BRI) and Integrated Services Digital Network Primary Rate Interface (ISDN-PRI).

Integrated Services Digital Network Basic Rate Interface (ISDN-BRI)
The interface between a communications system and terminal that includes two 64-kbps B-channels for transmitting voice or data and one 16-kbps D-channel for transmitting associated B-channel call control and out-of-band signaling information. ISDN-BRI also includes 48 kbps for transmitting framing and D-channel contention information, for a total interface speed of 192 kbps. ISDN-BRI serves ISDN terminals and digital terminals fitted with ISDN terminal adapters. See also Integrated Services Digital Network (ISDN) andIntegrated Services Digital Network Primary Rate Interface (ISDN-PRI).

Integrated Services Digital Network Primary Rate Interface (ISDN-PRI)
The interface between multiple communications systems that in North America includes 24 64-kbps channels, corresponding to the North American digital signal level-1 (DS1) standard rate of 1.544 Mbps. The most common arrangement of channels in ISDN-PRI is 23 64-kbps B-channels for transmitting voice and data and 1 64-kbps D-channel for transmitting associated B-channel call control and out-of-band signaling information. With nonfacility-associated signaling (NFAS), ISDN-PRI can include 24 B-channels and no D-channel. See alsoIntegrated Services Digital Network (ISDN) andIntegrated Services Digital Network Basic Rate Interface (ISDN-BRI).

intercept tone
A tone that indicates a dialing error or denial of the service requested.

interface
A common boundary between two systems or pieces of equipment.

internal call
A connection between two users within a system.

International Telecommunications Union (ITU)
Formerly known as International Telegraph and Telephone Consultative Committee (CCITT), ITU is an international organization that sets universal standards for data communications, including ISDN. ITU members are from telecommunications companies and organizations around the world. See alsoBX.25.

International Telegraph and Telephone Consultative Committee
See International Telecommunications Union (ITU).

interflow
The ability for calls to forward to other splits on the same PBX or a different PBX using the Call Forward All Calls feature.

intraflow
The ability for calls to redirect to other splits on the same PBX on a conditional or unconditional basis using call coverage busy, don't answer, or all criteria.

internal measurements
BCMS measurements that are made by the system. ACD measurements that are made external to the system (via External CMS) are referred to as external measurements.

in-use lamp
A red light on a multiappearance voice terminal that lights to show which call appearance will be selected when the handset is lifted or which call appearance is active when a user is off-hook.

INWATS
Inward Wide Area Telephone Service

IO
Information outlet

ISDN
See Integrated Services Digital Network (ISDN).

ISDN Gateway (IG)
A feature allowing integration of the switch and a host-based telemarketing application via a link to a gateway adjunct. The gateway adjunct is a 3B-based product that notifies the host-based telemarketing application of call events.

ISDN trunk
A trunk administered for use with ISDN-PRI. Also called ISDN facility.

ISDN-PRI terminal adapter
An interface between endpoint applications and an ISDN PRI facility. ISDN-PRI terminal adapters are currently available from other vendors and are primarily designed for video conferencing applications. Accordingly, currently available terminal adapters adapt the two pairs of video codec data (V.35) and dialing (RS-366) ports to an ISDN PRI facility.

IS/DTT
Integrated Services/digital tie trunk

ISN
Information Systems Network

ISO
International Standards Organization

ISV
Independent software vendor

ITP
Installation test procedure

ITU
International Telecommunications Union

IXC
Interexchange carrier code

K

kHz
Kilohertz

kbps
Kilobits per second

kbyte
Kilobyte

kg
Kilogram

L

LAN
Local area network

LAP-D
Link Access Procedure on the D-channel

LAPD
Link Access Procedure data

LATA
Local access and transport area

lb
Pound

LBO
Line buildout

LDN
Listed directory number

LDS
Long-distance service

LEC
Local exchange carrier

LED
See light-emitting diode (LED).

light-emitting diode (LED)
A semiconductor device that produces light when voltage is applied. LEDs provide a visual indication of the operational status of hardware components, the results of maintenance tests, the alarm status of circuit packs, and the activation of telephone features.

lightwave transceiver
Hardware that provides an interface to fiber-optic cable from port circuit packs and DS1 converter circuit packs. Lightwave transceivers convert electrical signals to light signals and vice versa.

line
A transmission path between a communications system or CO switching system and a voice terminal or other terminal.

line appearance
See appearance.

line buildout
A selectable output attenuation is generally required of DTE equipment because T1 circuits require the last span to lose 15–22.5 dB.

line port
Hardware that provides the access point to a communications system for each circuit associated with a telephone or data terminal.

link
A transmitter-receiver channel that connects two systems.

link-access procedure on the D-channel (LAPD)
A link-layer protocol on the ISDN-BRI and ISDN-PRI data-link layer (level 2). LAPD provides data transfer between two devices, and error and flow control on multiple logical links. LAPD is used for signaling and low-speed packet data (X.25 and mode 3) on the signaling (D-) channel and for mode-3 data communications on a bearer (B-) channel.

LINL
Local indirect neighbor link

local area network (LAN)
A networking arrangement designed for a limited geographical area. Generally, a LAN is limited in range to a maximum of 6.2 miles and provides high-speed carrier service with low error rates. Common configurations include daisy chain, star (including circuit-switched), ring, and bus.

logical link
The communications path between a processor and a BRI terminal.

loop-start trunk
A trunk on which, after establishing a connection with a distant switching system for an outgoing call, the system waits for a signal on the loop formed by the trunk leads before sending the digits of the called number.

loss plan
The overall plan, used in network design and management, for creating and maintaining consistent signal strength across the network. The term also applies to local management of signal strength to achieve appropriate levels for specific applications.

LSU
Local storage unit

LWC
Leave Word Calling

M

MAC
Medium access

MADU
: Modular asynchronous data unit

main distribution frame (MDF)
: A device that mounts to the wall inside the system equipment room. The MDF provides a connection point from outside telephone lines to the PBX switch and to the inside telephone stations.

main-satellite-tributary
: A private network configuration that can either stand alone or access an ETN. A main switch provides interconnection, via tie trunks, with one or more subtending switches, called satellites; all attendant positions for the main/satellite configuration; and access to and from the public network. To a user outside the complex, a main/satellite configuration appears as one switch, with one listed directory number (LDN). A tributary switch is connected to the main switch via tie trunks, but has its own attendant positions and LDN.

maintenance
: Activities involved in keeping a telecommunications system in proper working condition: the detection and isolation of software and hardware faults, and automatic and manual recovery from these faults.

management terminal
: The terminal that is used by the system administrator to administer the switch. The terminal may also be used to access the BCMS feature.

major alarm
: An indication of a failure that has caused critical degradation of service and requires immediate attention. Major alarms are automatically displayed on LEDs on the attendant console and maintenance or alarming circuit pack, logged to the alarm log, and reported to a remote maintenance facility, if applicable.

Manual-In work mode
: One of four agent work modes: the mode in which an agent is ready to process another call manually. *See* Auto-In Work mode for a contrast.

MAP
: Maintenance action process

MAPD
: Multiapplication platform for DEFINITY

MA-UUI
: Message-Associated User-to-User Signaling

Mbps
: Megabits per second

M-Bus
: Memory bus

Mbyte
: Megabyte

MCC
: Multicarrier cabinet

MCS
: Message Center Service

MCT
: Malicious Call Trace

MCU
Multipoint control unit

MDF
Main distribution frame

MDM
Modular data module

MDR
Message detail record

MEM
Memory

memory
A device into which information can be copied and held, and from which information can later be obtained.

memory shadowing link
An operating-system condition that provides a method for memory-resident programs to be more quickly accessed, allowing a system to reboot faster.

message center
An answering service that supplies agents to and stores messages for later retrieval.

message center agent
A member of a message-center hunt group who takes and retrieves messages for voice-terminal users.

MF
Multifrequency

MFB
Multifunction board

MFC
Multifrequency code

MHz
Megahertz

MIM
Management information message

minor alarm
An indication of a failure that could affect customer service. Minor alarms are automatically displayed on LEDs on the attendant console and maintenance or alarming circuit pack, sent to the alarm log, and reported to a remote maintenance facility, if applicable.

MIPS
Million instructions per second

MIS
Management information system

MISCID
Miscellaneous identification

MMCS
Multimedia Call Server

MMCH
: Multimedia call handling

MMI
: Multimedia interface

MMS
: Material Management Services

MO
: Maintenance object

modem
: A device that converts digital data signals to analog signals for transmission over telephone circuits. The analog signals are converted back to the original digital data signals by another modem at the other end of the circuit.

modem pooling
: A capability that provides shared conversion resources (modems and data modules) for cost-effective access to analog facilities by data terminals. When needed, modem pooling inserts a conversion resource into the path of a data call. Modem pooling serves both outgoing and incoming calls.

modular processor data module (MPDM)
: A processor data module (PDM) that can be configured to provide several kinds of interfaces (RS-232C, RS-449, and V.35) to customer-provided data terminal equipment (DTE). See also processor data module (PDM).

modular trunk data module (MTDM)
: A trunk data module that can be configured to provide several kinds of interfaces (RS-232, RS-449, and V.35) to customer-provided data terminal equipment.

modulator-demodulator
: See modem.

monitored call
: See active-notification call.

MOS
: Message-oriented signaling

MPDM
: Modular processor data module

MS
: Message server

ms
: Millisecond

MS/T
: Main satellite/tributary

MSG
: Message service

MSL
: Material stocking location

MSM
Modular System Management

MSS
Mass storage system

MSSNET
Mass storage/network control

MT
Management terminal

MTDM
Modular trunk data module

MTP
Maintenance tape processor

MTT
Multitasking terminal

multiappearance voice terminal
A terminal equipped with several call-appearance buttons for the same extension, allowing the user to handle more than one call on that same extension at the same time.

Multicarrier cabinet
A structure that holds one to five carriers. See also single-carrier cabinet.

Multifrequency Compelled (MFC) Release 2 (R2) signaling
A signal consisting of two frequency components, such that when a signal is transmitted from a switch, another signal acknowledging the transmitted signal is received by the switch. R2 designates signaling used in the United States and in countries outside the United States.

multiplexer
A device used to combine a number of individual channels into a single common bit stream for transmission.

multiplexing
A process whereby a transmission facility is divided into two or more channels, either by splitting the frequency band into a number of narrower bands or by dividing the transmission channel into successive time slots. See also time-division multiplexing (TDM).

multirate
The new N x DS0 service (see N x DS0).

MWL
Message-waiting lamp

N

N+1
Method of determining redundant backup requirements. Example: if four rectifier modules are required for a DC-powered single-carrier cabinet, a fifth rectifier module is installed for backup.

N x DS0

N x DS0, equivalently referred to as N x 64 kbps, is an emerging standard for wideband calls separate from H0, H11, and H12 ISDN channels. The emerging N x DS0 ISDN multirate circuit mode bearer service will provide circuit-switched calls with data-rate multiples of 64 kbps up to 1536 kbps on a T1 facility or up to 1920 kbps on an E1 facility. In the switch, N x DS0 channels will range up to 1984 kbps using NFAS E1 interfaces.

NANP

North American Numbering Plan

narrowband

A circuit-switched call at a data rate up to and including 64 kbps. All nonwideband switch calls are considered narrowband.

native terminal support

A predefined terminal type exists in switch software, eliminating the need to alias the terminal (that is, manually map call appearances and feature buttons onto some other natively supported terminal type).

NAU

Network access unit

NCA/TSC

Noncall-associated/temporary-signaling connection

NCOSS

Network Control Operations Support Center

NCSO

National Customer Support Organization

NEC

National Engineering Center

NEMA

National Electrical Manufacturer's Association

NETCON

Network-control circuit pack

network

A series of points, nodes, or stations connected by communications channels.

network-specific facility (NSF)

An information element in an ISDN-PRI message that specifies which public-network service is used. NSF applies only when Call-by-Call Service Selection is used to access a public-network service.

network interface

A common boundary between two systems in an interconnected group of systems.

NFAS

See Nonfacility-associated signaling (NFAS).

NI

Network interface

NID

Network Inward Dialing

NM
Network management

NN
National number

node
A switching or control point for a network. Nodes are either tandem (they receive signals and pass them on) or terminal (they originate or terminate a transmission path).

Nonfacility-associated signaling (NFAS)
A method that allows multiple T1 and/or E1 facilities to share a single D-channel to form an ISDN-PRI. If D-channel backup is not used, one facility is configured with a D-channel, and the other facilities that share the D-channel are configured without D-channels. If D-channel backup is used, two facilities are configured to have D-channels (one D-channel on each facility), and the other facilities that share the D-channels are configured without D-channels.

NPA
Numbering-plan area

NPE
Network processing element

NQC
Number of queued calls

NSE
Night-service extension

NSU
Network sharing unit

null modem cable
Special wiring of an RS-232-C cable such that a computer can talk to another computer (or to a printer) without a modem.

NXX
Public-network office code

O

OA
Operator assisted

occurrence
See appearance.

OCM
Outbound Call Management

offered load
The traffic that would be generated by all the requests for service occurring within a monitored interval, usually one hour.

ONS
On-premises station

OPS
Off-premises station

OPX
Off-premises extension

OQT
Oldest queued time

OSHA
Occupational Safety and Health Act

OSI
Open Systems Interconnect

OSS
Operations Support System

OSSI
Operational Support System Interface

OTDR
Optical time-domain reflectometer

othersplit
The work state that indicates that an agent is currently active on another split's call, or in ACW for another split.

OTL
Originating Test Line

OTQ
Outgoing trunk queuing

outgoing gateway
A PBX that routes an incoming call on a trunk administered for Supplementary Services Protocol B to a trunk *not* administered for Supplementary Services Protocol B.

P

PACCON
Packet control

packet
A group of bits (including a message element, which is the data, and a control information element (IE), which is the header) used in packet switching and transmitted as a discrete unit. In each packet, the message element and control IE are arranged in a specified format. See also **packet bus** and **packet switching**.

packet bus
A wide-bandwidth bus that transmits packets.

packet switching
A data-transmission technique whereby user information is segmented and routed in discrete data envelopes called packets, each with its own appended control information, for routing, sequencing, and error checking. Packet switching allows a channel to be occupied only during the transmission of a packet. On completion of the transmission, the channel is made available for the transfer of other packets. See also BX.25and **packet**.

PAD
Packet assembly/disassembly

paging trunk
A telecommunications channel used to access an amplifier for loudspeaker paging.

party/extension active on call
A party is on the call if he or she is actually connected to the call (in active talk or in held state). An originator of a call is always a party on the call. Alerting parties, busy parties, and tones are not parties on the call.

PBX
Private branch exchange

PC
See personal computer (PC).

PCM
See pulse-code modulation (PCM).

PCOL
Personal central-office line

PCOLG
Personal central-office line group

PCS
Permanent switched calls

PDM
See processor data module (PDM).

PDS
Premises Distribution System

PE
Processing element

PEC
Price element code

PEI
Processor element interchange

personal computer (PC)
A personally controllable microcomputer.

PGATE
Packet gateway

PGN
Partitioned group number

Phantom Calls
A feature that allows a call to originate either from a station AWOH or from a non-hunt group made up of AWOH stations.

PI
Processor interface

PIB
: Processor interface board

pickup group
: A group of individuals authorized to answer any call directed to an extension within the group.

PIDB
: Product image database

PKTINT
: Packet interface

PL
: Private line

PLS
: Premises Lightwave System

PMS
: Property Management System

PN
: Port network

PNA
: Private network access

POE
: Processor occupancy evaluation

POP
: Point of presence

port
: A data- or voice-transmission access point on a device that is used for communicating with other devices.

port carrier
: A carrier in a multicarrier cabinet or a single-carrier cabinet containing port circuit packs, power units, and service circuits. Also called a port cabinet in a single-carrier cabinet.

port network (PN)
: A cabinet containing a TDM bus and packet bus to which the following components are connected: port circuit packs, one or two tone-clock circuit packs, a maintenance circuit pack, service circuit packs, and (optionally) up to four expansion interface (EI) circuit packs in DEFINITY ECS. Each PN is controlled either locally or remotely by a switch processing element (SPE). See also expansion port network (EPN) and processor port network (PPN).

port-network connectivity
: The interconnection of port networks (PNs), regardless of whether the configuration uses direct or switched connectivity.

PPM
: 1. Parts per million
: 2. Periodic pulse metering

PPN
: See processor port network (PPN).

PRI
See Primary Rate Interface (PRI).

primary extension
The main extension associated with the physical voice or data terminal.

Primary Rate Interface (PRI)
A standard ISDN frame format that specifies the protocol used between two or more communications systems. PRI runs at 1.544 Mbps and, as used in North America, provides 23 64-kbps B-channels (voice or data) and one 64-kbps D-channel (signaling). The D-channel is the 24th channel of the interface and contains multiplexed signaling information for the other 23 channels.

PRI endpoint (PE)
The wideband switching capability introduces PRI endpoints on switch line-side interfaces. A PRI endpoint consists of one or more contiguous B-channels on a line-side T1 or E1 ISDN PRI facility and has an extension. Endpoint applications have call-control capabilities over PRI endpoints.

principal
A terminal that has its primary extension bridged on one or more other terminals.

principal (user)
A person to whom a telephone is assigned and who has message-center coverage.

private network
A network used exclusively for the telecommunications needs of a particular customer.

private network office code (RNX)
The first three digits of a 7-digit private network number.

processor carrier
See control carrier.

processor data module (PDM)
A device that provides an RS-232C DCE interface for connecting to data terminals, applications processors (APs), and host computers, and provides a DCP interface for connection to a communications system. See also modular processor data module (MPDM).

processor port network (PPN)
A port network controlled by a switch-processing element that is directly connected to that PN’s TDM bus and LAN bus. See also port network (PN).

processor port network (PPN) control carrier
A carrier containing the maintenance circuit pack, tone/clock circuit pack, and SPE circuit packs for a processor port network (PPN) and, optionally, port circuit packs.

PROCR
Processor

Property Management System (PMS)
A stand-alone computer used by lodging and health-services organizations for services such as reservations, housekeeping, and billing.

protocol
A set of conventions or rules governing the format and timing of message exchanges to control data movement and correction of errors.

PSC
Premises service consultant

PSDN
Packet-switch public data network

PT
Personal terminal

PTC
Positive temperature coefficient

PTT
Postal Telephone and Telegraph

public network
The network that can be openly accessed by all customers for local and long-distance calling.

pulse-code modulation (PCM)
An extension of pulse-amplitude modulation (PAM) in which carrier-signal pulses modulated by an analog signal, such as speech, are quantized and encoded to a digital, usually binary, format.

Q

QPPCN
Quality Protection Plan Change Notice

quadrant
A group of six contiguous DS0s in fixed locations on an ISDN-PRI facility. Note that this term comes from T1 terminology (one-fourth of a T1), but there are five quadrants on an E1 ISDN-PRI facility (30B + D).

queue
An ordered sequence of calls waiting to be processed.

queuing
The process of holding calls in order of their arrival to await connection to an attendant, to an answering group, or to an idle trunk. Calls are automatically connected in first-in, first-out sequence.

R

RAM
See random-access memory (RAM).

random-access memory (RAM)
A storage arrangement whereby information can be retrieved at a speed independent of the location of the stored information.

RBS
Robbed-bit signaling

RC
Radio controller

RCL
Restricted call list

read-only memory (ROM)
A storage arrangement primarily for information-retrieval applications.

recall dial tone
Tones signalling that the system has completed a function (such as holding a call) and is ready to accept dialing.

redirection criteria
Information administered for each voice terminal’s coverage path that determines when an incoming call is redirected to coverage.

Redirection on No Answer
An optional feature that redirects an unanswered ringing ACD call after an administered number of rings. The call is then redirected back to the agent.

release
To release a call is to initiate its disconnection.

release signal
The signal one switch sends to another to disconnect a call. If the calling switch ends the call, it sends a “forward” release signal. If the receiving switch ends the call, it sends a “backward” release signal.

remote home numbering-plan area code (RHNPA)
A foreign numbering-plan area code that is treated as a home area code by the Automatic Route Selection (ARS) feature. Calls can be allowed or denied based on the area code and the dialed CO code rather than just the area code. If the call is allowed, the ARS pattern used for the call is determined by these six digits.

Remote Operations Service Element (ROSE)
A CCITT and ISO standard that defines a notation and services that support interactions between the various entities that make up a distributed application.

REN
Ringer equivalency number

reorder tone
A tone to signal that at least one of the facilities, such as a trunk or a digit transmitter, needed for the call was not available.

report scheduler
Software that is used in conjunction with the system printer to schedule the days of the week and time of day that the desired reports are to be printed.

RFP
Request for proposal

RHNPA
See remote home numbering-plan area code (RHNPA).

RINL
Remote indirect neighbor link

RISC
Reduced-instruction-set computer

RLT
Release-link trunk

RMATS
Remote Maintenance, Administration, and Traffic System

RNX
Route-number index (private network office code)

ROM
See read-only memory (ROM).

ROSE
See Remote Operations Service Element (ROSE).

RPN
Routing-plan number

RS-232C
A physical interface specified by the Electronic Industries Association (EIA). RS-232C transmits and receives asynchronous data at speeds of up to 19.2 kbps over cable distances of up to 50 feet.

RS-449
Recommended Standard 449

RSC
Regional Support Center

S

S1
The first logical signalling channel of DCP. The channel is used to provide signaling information for DCP's I1 channel.

S2
The second logical signaling channel of DCP. The channel is used to provide signaling information for DCP's I2 channel.

SABM
Set Asynchronous Balance Mode

SAC
Send All Calls

SAKI
See sanity and control interface (SAKI).

sanity and control interface (SAKI)
A custom VLSI microchip located on each port circuit pack. The SAKI provides address recognition, buffering, and synchronization between the angel and the five control time slots that make up the control channel. The SAKI also scans and collects status information for the angel on its port circuit pack and, when polled, transmits this information to the archangel.

SAT
System access terminal

SBA
Simulated bridged appearance

SCC
1. See single-carrier cabinet.
2. Serial communications controller

SCD
Switch-control driver

SCI
Switch communications interface

SCO
System control office

SCOTCH
Switch Conferencing for TDM Bus in Concentration Highway

SCSI
See small computer system interface (SCSI).

SDDN
Software-Defined Data Network

SDI
Switched Digital International

SDLC
Synchronous data-link control

SDN
Software-defined network

SFRL
Single-frequency return loss

SID
Station-identification number

simplex system
A system that has no redundant hardware.

simulated bridged appearance
The same as a temporary bridged appearance; allows the terminal user (usually the principal) to bridge onto a call that had been answered by another party on his or her behalf.

single-carrier cabinet
A combined cabinet and carrier unit that contains one carrier. See also Multicarrier cabinet.

single-line voice terminal
A voice terminal served by a single-line tip and ring circuit (models 500, 2500, 7101A, 7103A).

SIT
Special-information tones

SLS
Service Level Supervisor

small computer system interface (SCSI)
An ANSI bus standard that provides a high-level command interface between host computers and peripheral devices.

SMDR
Station Message Detail Recording

SN
Switch Node

SNA
Systems Network Architecture

SNC
Switch Node Clock

SNI
Switch Node Interface

SNMP
Simple Network Management Protocol

software
A set of computer programs that perform one or more tasks.

SPE
Switch Processing Element

SPID
Service Profile Identifier

split
See ACD work mode.

split condition
A condition whereby a caller is temporarily separated from a connection with an attendant. A split condition automatically occurs when the attendant, active on a call, presses the start button.

split number
The split's identity to the switch and BCMS.

split report
A report that provides historical traffic information for internally measured splits.

split (agent) status report
A report that provides real-time status and measurement data for internally measured agents and the split to which they are assigned.

SSI
Standard serial interface

SSM
Single-site management

SSV
Station service

ST3
Stratum 3 clock board

staffed
Indicates that an agent position is logged in. A staffed agent functions in one of four work modes: Auto-In, Manual-In, ACW, or AUX-Work.

STARLAN
Star-Based Local Area Network

Station Message Detail Recording (SMDR)
An obsolete term now called CDR — a switch feature that uses software and hardware to record call data. See Call Detail Recording (CDR).

standard serial interface (SSI)
A communications protocol developed for use with 500-type business communications terminals (BCTs) and 400-series printers.

status lamp
A green light that shows the status of a call appearance or a feature button by the state of the light (lit, flashing, fluttering, broken flutter, or unlit).

stroke counts
A method used by ACD agents to record up to nine customer-defined events per call when CMS is active.

SVN
Security-violation notification

switch
Any kind of telephone switching system. See also communications system.

switchhook
The buttons located under the receiver on a voice terminal.

switch-node (SN) carrier
A carrier containing a single switch node, power units, and, optionally, one or two DS1 converter circuit packs. An SN carrier is located in a center-stage switch.

switch-node (SN) clock
The circuit pack in an SN carrier that provides clock and maintenance alarm functions and environmental monitors.

switch-node interface (SNI)
The basic building block of a switch node. An SNI circuit pack controls the routing of circuit, packet, and control messages.

switch-node link (SNL)
The hardware that provides a bridge between two or more switch nodes. The SNL consists of the two SNI circuit packs residing on the switch nodes and the hardware connecting the SNIs. This hardware can include lightwave transceivers that convert the SNI's electrical signals to light signals, the copper wire that connects the SNIs to the lightwave transceivers, a full-duplex fiber-optic cable, DS1 converter circuit cards and DS1 facilities if a company does not have rights to lay cable, and appropriate connectors.

switch-processing element (SPE)
A complex of circuit packs (processor, memory, disk controller, and bus-interface cards) mounted in a PPN control carrier. The SPE serves as the control element for that PPN and, optionally, for one or more EPNs.

SXS
Step-by-step

synchronous data transmission
A method of sending data in which discrete signal elements are sent at a fixed and continuous rate and specified times. See also association.

SYSAM
System Access and Administration

system administrator

The person who maintains overall customer responsibility for system administration. Generally, all administration functions are performed from the Management Terminal. The switch requires a special login, referred to as the system administrator login, to gain access to system-administration capabilities.

system printer

An optional printer that may be used to print scheduled reports via the report scheduler.

system report

A report that provides historical traffic information for internally measured splits.

system-status report

A report that provides real-time status information for internally measured splits.

system manager

A person responsible for specifying and administering features and services for a system.

system reload

A process that allows stored data to be written from a tape into the system memory (normally after a power outage).

T

T1

A digital transmission standard that in North America carries traffic at the DS1 rate of 1.544 Mbps. A T1 facility is divided into 24 channels (DS0s) of 64 kbps. These 24 channels, with an overall digital rate of 1.536 Mbps, and an 8-kbps framing and synchronization channel make up the 1.544-Mbps transmission. When a D-channel is present, it occupies channel 24. T1 facilities are also used in Japan and some Middle-Eastern countries.

TAAS

Trunk Answer from Any Station

TABS

Telemetry asynchronous block serial

TAC

Trunk-access code

tandem switch

A switch within an electronic tandem network (ETN) that provides the logic to determine the best route for a network call, possibly modifies the digits outpulsed, and allows or denies certain calls to certain users.

tandem through

The switched connection of an incoming trunk to an outgoing trunk without human intervention.

tandem tie-trunk network (TTTN)

A private network that interconnects several customer switching systems.

TC

Technical consultant

TCM

Traveling class mark

TDM
See time-division multiplexing (TDM).

TDR
Time-of-day routing

TEG
Terminating extension group

terminal
A device that sends and receives data within a system. See also administration terminal.

tie trunk
A telecommunications channel that directly connects two private switching systems.

time-division multiplex (TDM) bus
A bus that is time-shared regularly by preallocating short time slots to each transmitter. In a PBX, all port circuits are connected to the TDM bus, permitting any port to send a signal to any other port.

time-division multiplexing (TDM)
Multiplexing that divides a transmission channel into successive time slots. See also multiplexing.

time interval
The period of time, either one hour or one-half hour, that BCMS measurements are collected for a reports.

time slice
See **time interval**.

time slot
64 kbps of digital information structured as eight bits every 125 microseconds. In the switch, a time slot refers to either a DS0 on a T1 or E1 facility or a 64-kbps unit on the TDM bus or fiber connection between port networks.

time slot sequence integrity
The situation whereby the N octets of a wideband call that are transmitted in one T1 or E1 frame arrive at the output in the same order that they were introduced.

to control
An application can invoke *Third Party Call Control* capabilities using either an adjunct-control or domain-control association.

to monitor
An application can receive *event reports* on an active-notification, adjunct-control, or domain-control association.

TOD
Time of day

tone ringer
A device with a speaker, used in electronic voice terminals to alert the user.

TOP
Task-oriented protocol

trunk
A dedicated telecommunications channel between two communications systems or COs.

trunk allocation
The manner in which trunks are selected to form wideband channels.

trunk-data module

A device that connects off-premises private-line trunk facilities and DEFINITY ECS. The trunk-data module converts between the RS-232C and the DCP, and can connect to DDD modems as the DCP member of a modem pool.

trunk group

Telecommunications channels assigned as a group for certain functions that can be used interchangeably between two communications systems or COs.

TSC

Technical Service Center

TTI

Terminal translation initialization

TTR

Touch-tone receiver

TTT

Terminating trunk transmission

TTTN

See tandem tie-trunk network (TTTN).

TTY

Teletypewriter

U

UAP

Usage-allocation plan

UART

Universal asynchronous transmitter

UCD

Uniform call distribution

UCL

Unrestricted call list

UDP

1. User Datagram Protocol - Transport layer; connectionless, unreliable, fast.

2. Uniform Dial Plan -A feature that allows a unique 4- or 5-digit number assignment for each terminal in a multiswitch configuration such as a DCS or main-satellite-tributary system.

Underwriter Laboratories

UM

User manager

UNMA

Unified Network Management Architecture

UNP
Uniform numbering plan

UPS
Uninterruptible power supply

USOP
User service-order profile

UUCP
UNIX-to-UNIX Communications Protocol

UUI
User-to-user information

V

VAR
Value-added reseller

VDN
See vector directory number (VDN).

vector directory number (VDN)
An extension that provides access to the Vectoring feature on the switch. Vectoring allows a customer to specify the treatment of incoming calls based on the dialed number.

vector-controlled split
A hunt group or ACD split administered with the vector field enabled. Access to such a split is possible only by dialing a VDN extension.

VIS
Voice Information System

VLSI
Very-large-scale integration

VM
Voltmeter

VNI
Virtual nodepoint identifier

VOA
VDN of origin announcement

voice terminal
A single-line or multiappearance telephone.

W

WATS
See Wide Area Telecommunications Service (WATS).

WCC
World-Class Core

WCR
World-Class Routing

WCTD
World-Class Tone Detection

WFB
Wireless fixed base

Wide Area Telecommunications Service (WATS)
A service in the United States that allows calls to certain areas for a flat-rate charge based on expected usage.

wideband
A circuit-switched call at a data rate greater than 64 kbps. A circuit-switched call on a single T1 or E1 facility with a bandwidth between 128 and 1536 (T1) or 1984 (E1) kbps in multiples of 64 kbps. H0, H11, H12, and N x DS0 calls are wideband.

wideband access endpoint
Access endpoints, extended with wideband switching to include wideband access endpoints. A wideband access endpoint consists of one or more contiguous DS0s on a line-side T1 or E1 facility and has an extension. The Administered Connections feature provides call control for calls originating from wideband access endpoints.

wink-start tie trunk
A trunk with which, after making a connection with a distant switching system for an outgoing call, the system waits for a momentary signal (wink) before sending the digits of the called number. Similarly, on an incoming call, the system sends the wink signal when ready to receive digits.

work mode
One of four states (Auto-In, Manual-In, ACW, AUX-Work) that an ACD agent can be in. Upon logging in, an agent enters AUX-Work mode. To become available to receive ACD calls, the agent enters Auto-In or Manual-In mode. To do work associated with a completed ACD call, an agent enters ACW mode.

work state
An ACD agent may be a member of up to three different splits. Each ACD agent continuously exhibits a work state for every split of which it is a member. Valid work states are Avail, Unstaffed, AUX-Work, ACW, ACD (answering an ACD call), ExtIn, ExtOut, and OtherSpl. An agent’s work state for a particular split may change for a variety of reasons (example: when a call is answered or abandoned, or the agent changes work modes). The BCMS feature monitors work states and uses this information to provide BCMS reports.

write operation
The process of putting information onto a storage medium, such as a hard disk.

WSA
Waiting session accept

WSS
Wireless Subscriber System

Z

ZCS

Zero Code Suppression

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Numerics

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