



Feature Description and Implementation for Avaya Communication Manager

555-245-205
Issue 1
June 2004

**Copyright 2004, Avaya Inc.
All Rights Reserved**

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.

Warranty

Avaya Inc. provides a limited warranty on this product. Refer to your sales agreement to establish the terms of the limited warranty. In addition, Avaya's standard warranty language as well as information regarding support for this product, while under warranty, is available through the following Web site: <http://www.avaya.com/support>.

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.

Disclaimer

Avaya is not responsible for any modifications, additions or deletions to the original published version of this documentation unless such modifications, additions or deletions were performed by Avaya. Customer and/or End User agree to indemnify and hold harmless Avaya, Avaya's agents, servants and employees against all claims, lawsuits, demands and judgments arising out of, or in connection with, subsequent modifications, additions or deletions to this documentation to the extent made by the Customer or End User.

How to Get Help

For additional support telephone numbers, go to the Avaya support Web site: <http://www.avaya.com/support>. If you are:

- Within the United States, click the *Escalation Management* link. Then click the appropriate link for the type of support you need.
- Outside the United States, click the *Escalation Management* link. Then click the *International Services* link that includes telephone numbers for the international 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

TCP/IP Facilities

Customers may experience differences in product performance, reliability and security depending upon network configurations/design and topologies, even when the product performs as warranted.

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.

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, or IEC 60950-1, 1st Edition, including all relevant national deviations as listed in Compliance with IEC for Electrical Equipment (IECEE) CB-96A.

Safety of Information Technology Equipment, CAN/CSA-C22.2 No. 60950-00 / UL 60950, 3rd Edition, or CAN/CSA-C22.2 No. 60950-1-03 / UL 60950-1.

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.

The equipment described in this document may contain Class 1 LASER Device(s). These devices comply with the following standards:

- EN 60825-1, Edition 1.1, 1998-01
- 21 CFR 1040.10 and CFR 1040.11.

The LASER devices used in Avaya equipment typically operate within the following parameters:

Typical Center Wavelength	Maximum Output Power
830 nm - 860 nm	-1.5 dBm
1270 nm - 1360 nm	-3.0 dBm
1540 nm - 1570 nm	5.0 dBm

Luokan 1 Laserlaite

Klass 1 Laser Apparat

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

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

Power Line Emissions, IEC 61000-3-2: Electromagnetic compatibility (EMC) – Part 3-2: Limits – Limits for harmonic current emissions.

Power Line Emissions, IEC 61000-3-3: Electromagnetic compatibility (EMC) – Part 3-3: Limits – Limitation of voltage changes, voltage fluctuations and flicker in public low-voltage supply systems.

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.

REN Number

For MCC1, SCC1, CMC1, G600, and G650 Media Gateways:

This equipment complies with Part 68 of the FCC rules. On either the rear or inside the front cover 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.

For G350 and G700 Media Gateways:

This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. On the rear of this equipment is a label that contains, among other information, a product identifier in the format US:AAAEQ##TXXXX. The digits represented by ## are the ringer equivalence number (REN) without a decimal point (for example, 03 is a REN of 0.3). If requested, this number must be provided to the telephone company.

For all media gateways:

The REN is used to determine the quantity of devices that 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 tables.

For MCC1, SCC1, CMC1, G600, and G650 Media Gateways:

Manufacturer's Port Identifier	FIC Code	SOC/REN/ A.S. Code	Network Jacks
Off premises station	OL13C	9.0F	RJ2GX, RJ21X, RJ11C
DID trunk	02RV2-T	0.0B	RJ2GX, RJ21X
CO trunk	02GS2	0.3A	RJ21X
	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	6.0F	RJ48C, RJ48M
	04DU9-IKN	6.0F	RJ48C, RJ48M
	04DU9-ISN	6.0F	RJ48C, RJ48M
120A4 channel service unit	04DU9-DN	6.0Y	RJ48C

For G350 and G700 Media Gateways:

Manufacturer's Port Identifier	FIC Code	SOC/REN/A.S. Code	Network Jacks
Ground Start CO trunk	02GS2	1.0A	RJ11C
DID trunk	02RV2-T	AS.0	RJ11C
Loop Start CO trunk	02LS2	0.5A	RJ11C
1.544 digital interface	04DU9-BN	6.0Y	RJ48C
	04DU9-DN	6.0Y	RJ48C
	04DU9-IKN	6.0Y	RJ48C
	04DU9-ISN	6.0Y	RJ48C
Basic Rate Interface	02IS5	6.0F	RJ49C

For all media gateways:

If the terminal equipment (for example, the media server or media gateway) 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.

A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA. A compliant telephone cord and modular plug is provided with this product. It is designed to be connected to a compatible modular jack that is also compliant. 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 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.

Installation and Repairs

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be coordinated by a representative designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect 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://www.avaya.com/support>.

All Avaya media servers and media gateways 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 Européenne*) 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).

Copies of these Declarations of Conformity (DoCs) can be obtained by contacting your local sales representative and are available on the following Web site: <http://www.avaya.com/support>.

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。

To order copies of this and other documents:

Call: Avaya Publications Center
Voice 1.800.457.1235 or 1.207.866.6701
FAX 1.800.457.1764 or 1.207.626.7269

Write: Globalware Solutions
200 Ward Hill Avenue
Haverhill, MA 01835 USA
Attention: Avaya Account Management

E-mail: totalware@gwsmail.com

For the most current versions of documentation, go to the Avaya support Web site: <http://www.avaya.com/support>.

Contents

Abbreviated Dialing	63
• Detailed description of Abbreviated Dialing	63
• Hardware requirements for Abbreviated Dialing	64
• Administering Abbreviated Dialing	64
Prerequisites for Abbreviated Dialing	64
Screens for administering Abbreviated Dialing	65
Adding Abbreviated Dialing lists	65
Assigning telephones for group lists	66
• End-user procedures for Abbreviated Dialing	67
Using the Abbreviated Dialing program feature	67
• Reports for Abbreviated Dialing	67
• Considerations for Abbreviated Dialing	68
• Interactions for Abbreviated Dialing	68
• Troubleshooting Abbreviated Dialing	69
 Access Security Gateway	 71
• Detailed Description of Access Security Gateway	71
• Hardware requirements for Access Security Gateway	71
• Administering Access Security Gateway	71
Prerequisites for administering Access Security Gateway	72
Screens for administering Access Security Gateway	72
Setting up Access Security Gateway	72
Disabling Access Security Gateway	74
Restarting Access Security Gateway	75
Modifying Access Security Gateway for a lost key	75
Monitoring the Access Security Gateway History Log	75
• Reports for Access Security Gateway	76
• Considerations for Access Security Gateway	77
• Interactions for Access Security Gateway	77
 Administration Change Notification	 79
• Detailed Description of Administration Change Notification	79
• Hardware requirements for Administration Change Notification	79

• Administering Administration Change Notification	80
Prerequisites for administering Administration Change Notification	80
Screens for administering Administration Change Notification	80
Initiating Administration Change Notification	80
• Reports for Administration Change Notification	81
• Considerations for Administration Change Notification	81
• Interactions for Administration Change Notification	81
Administered Connections	83
• Detailed description of Administered Connections	83
Access endpoints	83
Typical applications	83
Establishing Administered Connections	84
Dropping Administered Connections	85
Administered Connections failure: auto restoration and fast retry	85
• Hardware requirements for Administered Connections	86
• Administering Administered Connections	86
Prerequisites for administering Administered Connections	86
Screens for administering Administered Connections	86
Setting up Administered Connections	87
• Reports for Administered Connections	87
• Considerations for Administered Connections	88
• Interactions for Administered Connections	88
Administrable Language Displays	91
• Detailed description of Administrable Language Displays	91
• Hardware requirements for Administrable Language Displays	92
• Administering Language Displays	92
Prerequisites for administering Administrable Language Displays	92
Screens for administering Administrable Language Displays	93
Setting the display language	93
Entering translations for a user-defined language	94
Administering Unicode display	95

• Reports for Administrable Language Displays	96
• Considerations for Administrable Language Displays	96
• Interactions for Administrable Language Displays	96
• Troubleshooting Administrable Language Displays	97

Administration Without Hardware 99

• Detailed description of Administration Without Hardware	99
Physical characteristics of an AWOH telephone	99
User activated features	99
Association and disassociation	99
Phantom extensions	100
• Hardware requirements for Administration Without Hardware	100
• Administering Administration Without Hardware	100
Prerequisites for administering Administration Without Hardware	100
Screens for administering Administration Without Hardware	101
Assigning AWOH for a hunt-group queue	101
Assigning AWOH to a telephone	102
Assigning AWOH to an attendant console	103
Assigning AWOH to a data module	103
• Reports for Administration Without Hardware	104
• Considerations for Administration Without Hardware	104
• Interactions for Administration Without Hardware	105

Alphanumeric Dialing 113

• Detailed description of Alphanumeric Dialing	113
• Hardware requirements for Alphanumeric Dialing	113
• Administering Alphanumeric Dialing	113
Screens for administering Alphanumeric Dialing	114
• Reports for Alphanumeric Dialing	114
• Considerations for Alphanumeric Dialing	114
• Interactions for Alphanumeric Dialing	114

Announcements	115
• Detailed description of Announcements	115
Announcement sources for the Avaya G700 Media Gateway	116
VAL announcements	116
VAL Manager	117
Local announcements on the G700 Media Gateway	117
Integrated announcements	117
Single integrated announcement circuit packs	118
Multiple integrated announcement circuit packs	118
Backing up circuit packs	118
Compression rates	119
Barging-in on announcements	119
• Hardware requirements for Announcements	119
• Administering Announcements	120
Prerequisites for administering Announcements	120
Screens for administering Announcements	120
Displaying all announcements	122
Adding announcement data modules	123
Adding announcement extensions	123
Recording and changing announcements	124
Saving announcements	126
Copying announcements	126
Restoring announcements	126
Deleting and erasing announcements	127
Setting up continuous-play announcements	127
Recording VAL announcements	128
Converting announcement files to VAL format	129
Converting announcements for the Avaya Interactive Voice Response (IVR)	130
Deleting VAL announcements	131
Using the SAT to delete individual VAL announcement files	131
Using the SAT to delete all VAL announcements on a circuit pack	131
Using FTP to Manage VAL announcements	132
Setting up an FTP session	132
Performing tasks in an FTP session	134
Recording announcements for TTY callers	137

• Reports for Announcements	138
• Considerations for Announcements	138
• Interactions for Announcements	139
• Troubleshooting announcements	140
Attendant Auto-Manual Splitting	141
• Detailed description of Attendant Auto-Manual Splitting	141
• Hardware requirements for Attendant Auto-Manual Splitting	141
• Administering Attendant Auto-Manual Splitting	141
Prerequisites for administering Attendant Auto-Manual Splitting	142
Screens for administering Attendant Auto-Manual Splitting	142
• Reports for Attendant Auto-Manual Splitting	142
• Considerations for Attendant Auto-Manual Splitting	142
• Interactions for Attendant Auto-Manual Splitting	142
Attendant Auto Start and Don't Split	143
• Detailed description of Attendant Auto Start and Don't Split	143
• Hardware requirements for Attendant Auto Start and Don't Split	143
• Administering Attendant Auto Start and Don't Split	144
Prerequisites for administering Attendant Auto Start and Don't Split	144
Screens for administering Attendant Auto Start and Don't Split	144
Assigning a Don't Split button	144
• Reports for Attendant Auto Start and Don't Split	145
• Considerations for Attendant Auto Start and Don't Split	145
• Interactions for Attendant Auto Start and Don't Split	146
Attendant Backup	147
• Detailed description of Attendant Backup	147
Attendant Backup Alerting	148
• Hardware requirements for Attendant Backup	148
• Administering Attendant Backup	148
Prerequisites for administering Attendant Backup	148
Screens for administering Attendant Backup	149

Setting up Attendant Backup telephones	149
Configuring your system	149
Defining Class of Service console permissions	151
Assigning console permissions to backup telephones	151
Training designated users	153
• End-user procedures for Attendant Backup	153
• Reports for Attendant Backup	153
• Considerations for Attendant Backup	154
• Interactions for Attendant Backup	154
Attendant Call Waiting	155
• Detailed description of Attendant Call Waiting	155
• Hardware requirements for Attendant Call Waiting	155
• Administering Attendant Call Waiting	156
Prerequisites for administering Attendant Call Waiting	156
Screens for administering Attendant Call Waiting	156
Setting up single-line telephones	157
Changing the call-waiting signal	157
Modifying timed intervals	158
• Reports for Attendant Call Waiting	159
• Considerations for Attendant Call Waiting	159
• Interactions for Attendant Call Waiting	160
Attendant Calling of Inward Restricted Stations	161
• Detailed description of Attendant Calling of Inward Restricted Stations	161
• Hardware requirements for Attendant Calling of Inward Restricted Stations	161
• Administering Attendant Calling of Inward Restricted Stations	161
Prerequisites for administering Attendant Calling of Inward Restricted Stations	162
Screens for administering Attendant Calling of Inward Restricted Stations	162
Setting up Class of Restriction override	162
• Reports for Attendant Calling of Inward Restricted Stations	162

• Considerations for Attendant Calling of Inward Restricted Stations	162
• Interactions for Attendant Calling of Inward Restricted Stations	163
Attendant Conference	165
• Detailed description of Attendant Conference	165
• Hardware requirements for Attendant Conference	165
• Administering Attendant Conference	165
Prerequisites for administering Attendant Conference	165
Screens for administering Attendant Conference	166
Setting up Attendant Conference	166
• Reports for Attendant Conference	167
• Considerations for Attendant Conference	167
• Interactions for Attendant Conference	168
Attendant Control of Trunk Group Access	169
• Detailed description of Attendant Control of Trunk Group Access	169
• Hardware requirements for Attendant Control of Trunk Group Access	170
• Administering Attendant Control of Trunk Group Access	170
Prerequisites for administering Attendant Control of Trunk Group Access	170
Screens for administering Attendant Control of Trunk Group Access	171
Setting the trunk group threshold	171
Assigning Attendant Control of Trunk Group Access buttons	172
• Reports for Attendant Control of Trunk Group Access	172
• Considerations for Attendant Control of Trunk Group Access	173
• Interactions for Attendant Control of Trunk Group Access	173
Attendant Direct Extension Selection	175
• Detailed description of Attendant Direct Extension Selection	175
Standard DXS Tracking	176
Enhanced DXS Tracking	176
Group Display button	176

• Hardware requirements for Attendant Direct Extension Selection	176
• Administering Attendant Direct Extension Selection	177
Prerequisites for administering Attendant Direct Extension Selection	177
Screens for administering Attendant Direct Extension Selection	177
• Reports for Attendant Direct Extension Selection	177
• Considerations for Attendant Direct Extension Selection	178
• Interactions for Attendant Direct Extension Selection	178
Attendant Direct Trunk Group Selection	179
• Detailed description of Attendant Direct Trunk Group Selection	179
• Hardware requirements for Attendant Direct Trunk Group Selection	179
• Administering Attendant Direct Trunk Group Selection	180
Prerequisites for administering Attendant Direct Trunk Group Selection	180
Screens for administering Attendant Direct Extension Selection	180
• Reports for Attendant Direct Trunk Group Selection	180
• Considerations for Attendant Direct Trunk Group Selection	181
• Interactions for Attendant Direct Trunk Group Selection	181
Attendant Intrusion	183
• Detailed description of Attendant Intrusion	183
• Hardware requirements for Attendant Intrusion	183
• Administering Attendant Intrusion	183
Prerequisites for administering Attendant Intrusion	183
Screens for administering Attendant Intrusion	184
Assigning an intrusion button	184
• Reports for Attendant Intrusion	184
• Considerations for Attendant Intrusion	185
• Interactions for Attendant Intrusion	185

Attendant Lockout – Privacy 187

- Detailed description of Attendant Lockout – Privacy 187
- Hardware requirements for Attendant Lockout – Privacy 187
- Administering Attendant Lockout – Privacy 187
 - Prerequisites for administering Attendant Lockout – Privacy 188
 - Screens for administering Attendant Lockout – Privacy 188
 - Activating or deactivating the Attendant Lockout – Privacy feature 188
- Reports for Attendant Lockout – Privacy 189
- Considerations for Attendant Lockout – Privacy 189
- Interactions for Attendant Lockout – Privacy 190

Attendant Override of Diversion Features 191

- Detailed description of Attendant Override of Diversion Features 191
- Hardware requirements for Attendant Override of Diversion Features 191
- Administering Attendant Override of Diversion Features 191
 - Prerequisites for administering Attendant Override of Diversion Features 191
 - Screens for administering Attendant Override of Diversion Features 192
- Reports for Attendant Override of Diversion Features 192
- Considerations for Attendant Override of Diversion Features 192
- Interactions for Attendant Override of Diversion Features 192

Attendant Priority Queue 193

- Detailed description of Attendant Priority Queue 193
 - Priority by Call Category 193
 - Priority by Call Type 194
- Hardware requirements for Attendant Priority Queue 195
- Administering Attendant Priority Queue 195
 - Prerequisites for administering Attendant Priority Queue 195
 - Screens for administering Attendant Priority Queue 196
 - Setting category priorities 196
 - Setting the number of calls in the queue 197

Assigning a Call Type button	198
Translating the Call Type button into a user-defined language	198
• Reports for Attendant Priority Queue	199
• Considerations for Attendant Priority Queue	199
• Interactions for Attendant Priority Queue	199
Attendant Recall	201
• Detailed description of Attendant Recall	201
• Hardware requirements for Attendant Recall	201
• Administering Attendant Recall	201
Prerequisites for administering Attendant Recall	201
Screens for administering Attendant Recall	202
• End-user procedures for Attendant Recall	202
• Reports for Attendant Recall	202
• Considerations for Attendant Recall	202
• Interactions for Attendant Recall	202
Attendant Serial Calling	203
• Detailed description of Attendant Serial Calling	203
• Hardware requirements for Attendant Serial Calling	203
• Administering Attendant Serial Calling	203
Prerequisites for administering Attendant Serial Calling	203
Screens for administering Attendant Serial Calling	204
• Reports for Attendant Serial Calling	204
• Considerations for Attendant Serial Calling	204
• Interactions for Attendant Serial Calling	204
Attendant Split Swap	205
• Detailed description of Attendant Split Swap	205
• Hardware requirements for Attendant Split Swap	205
• Administering Attendant Split Swap	205
Prerequisites for administering Attendant Split Swap	205
Screens for administering Attendant Split Swap	206
Assigning a split-swap button	206
• Reports for Attendant Split Swap	207

- Considerations for Attendant Split Swap 207
- Interactions for Attendant Split Swap 207

Attendant Timers 209

- Detailed description of Attendant Timers 209
- Hardware requirements for Attendant Timers 210
- Administering Attendant Timers 210
 - Prerequisites for administering Attendant Timers 210
 - Screens for administering Attendant Timers 211
 - Setting up Attendant Timers 211
- Reports for Attendant Timers 212
- Considerations for Attendant Timers 212
- Interactions for Attendant Timers 213

Attendant Trunk Identification 215

- Detailed description of Attendant Trunk Identification 215
- Hardware requirements for Attendant Trunk Identification 215
- Administering Attendant Trunk Identification 215
 - Prerequisites for administering Attendant Trunk Identification 215
 - Screens for administering Attendant Trunk Identification 216
- Reports for Attendant Trunk Identification 216
- Considerations for Attendant Trunk Identification 216
- Interactions for Attendant Trunk Identification 216

Audible Message Waiting 217

- Detailed description of Audible Message Waiting 217
- Hardware requirements for Audible Message Waiting 217
- Administering Audible Message Waiting 218
 - Prerequisites for administering Audible Message Waiting 218
 - Screens for administering Audible Message Waiting 218
 - Administering Audible Message Waiting for a user 218
- Reports for Audible Message Waiting 219
- Considerations for Audible Message Waiting 219
- Interactions for Audible Message Waiting 220

Attendant Vectoring	221
• Detailed description of Attendant Vectoring	221
• Hardware requirements for Attendant Vectoring	221
• Administering Attendant Vectoring	221
Prerequisites for administering Attendant Vectoring	222
Screens for administering Attendant Vectoring	223
Creating a VDN extension for Attendant Vectoring	223
Assigning the VDN extension for Attendant Vectoring to a console	224
Assigning the VDN extension for Attendant Vectoring to a tenant	225
• Reports for Attendant Vectoring	226
• Considerations for Attendant Vectoring	226
• Interactions for Attendant Vectoring	226
 AUDIX One-Step Recording	 227
• Detailed description of AUDIX One-Step Recording	227
Feature button	227
Language options	227
Periodic alerting tone	228
Ready indication tone	228
Recording delay timer	228
• Hardware requirements for AUDIX One-Step Recording	229
• Administering AUDIX One-Step Recording	229
Prerequisites for administering AUDIX One-Step Recording	229
Screens for administering AUDIX One-Step Recording	230
Assigning AUDIX One-Step Recording parameters	230
Translating telephone feature buttons and labels	232
Translating the AUDIX One-Step Recording feature display to a user-defined language	232
Translating the AUDIX One-Step Recording button label to a user-defined language	233
Assigning the feature button to a telephone	234
Changing the characteristics of the zip tone	234
• End-user procedures for AUDIX One-Step Recording	237
• Reports for AUDIX One-Step Recording	237
• Considerations for AUDIX One-Step Recording	238

- Interactions for AUDIX One-Step Recording 239
- Troubleshooting AUDIX One-Step Recording 240

Automated Attendant 243

- Detailed description of Automated Attendant 243
- Hardware requirements for Automated Attendant 243
- Administering Automated Attendant 243
 - Prerequisites for administering Automated Attendant 244
 - Screens for administering Automated Attendant 244
 - Setting the prompting timeout 245
 - Administering a vector directory number 245
 - Administering announcements 245
 - Controlling hunt groups by vector 246
 - Administering call vectors 246
 - Assigning a caller information button on a multiappearance telephone 246
 - Assigning a caller information button on an attendant console 247
- Reports for Automated Attendant 248
- Considerations for Automated Attendant 248
- Interactions for Automated Attendant 249

Automatic Callback 251

- Detailed description of Automatic Callback 251
 - Ringback Queing 252
 - Analog Busy Automatic Callback Without Flash 252
- Hardware requirements for Automatic Callback 252
- Administering Automatic Callback 253
 - Prerequisites for administering Automatic Callback 253
 - Screens for administering Automatic Callback 253
 - Assigning a feature access code (FAC) for Automatic Callback 254
 - Setting the no-answer timeout interval 254
 - Assigning a feature button for Automatic Callback 255
 - Setting the queue length for Ringback Queing 256
- Reports for Automatic Callback 257
- Considerations for Automatic Callback 258
- Interactions for Automatic Callback 259

Automatic Circuit Assurance	261
• Detailed description of Automatic Circuit Assurance	261
The referral call	261
The audit trail	262
• Hardware requirements for Automatic Circuit Assurance	262
• Administering Automatic Circuit Assurance	262
Screens for administering Automatic Circuit Assurance	262
• Reports for Automatic Circuit Assurance	263
• Considerations for Automatic Circuit Assurance	263
• Interactions for Automatic Circuit Assurance	263
 Automatic Number Identification	 265
• Detailed description of Automatic Number Identification	265
Outgoing Automatic Number Identification	265
• Hardware requirements for Automatic Number Identification	265
• Administering Automatic Number Identification	266
• Reports for Automatic Number Identification	266
• Considerations for Automatic Number Identification	266
• Interactions for Automatic Number Identification	266
 Authorization Codes	 269
• Detailed description of Authorization Codes	269
Example	269
Length of authorization codes	269
Using authorization codes	270
AAR and ARS calls	270
• Hardware requirements for Authorization Codes	271
• Administering Authorization Codes	271
Prerequisites for administering Authorization Codes	271
Screens for administering Authorization Codes	271
Setting up Authorization Codes	272
Creating Authorization Codes with a specific Class of Restriction (COR)	274
• Reports for Authorization Codes	275
• Considerations for Authorization Codes	275
• Interactions for Authorization Codes	276

Bridged Call Appearance 277

- Detailed description of Bridged Call Appearance 277
 - Sample uses 278
 - Administrable buttons and lamps for multiappearance telephones 278
- Hardware requirements for Bridged Call Appearance 278
- Administering Bridged Call Appearance 279
 - Prerequisites for administering Bridged Call Appearance 279
 - Screens for administering Bridged Call Appearance 279
 - Creating a bridged call appearance on a single-line telephone 280
 - Creating a bridged call appearance on a multiappearance telephone 281
- Reports for Bridged Call Appearance 283
- Considerations for Bridged Call Appearance 283
- Interactions for Bridged Call Appearance 284

Bulletin Board 293

- Detailed description of Bulletin Board 293
- Hardware requirements for Bulletin Board 293
- Administering Bulletin Board 294
 - Prerequisites for administering Bulletin Board 294
 - Changing bulletin board information 295
 - Screens for administering Bulletin Board 297
- Reports for Bulletin Board 297
- Considerations for Bulletin Board 298
- Interactions for Bulletin Board 298

Busy Indicator 299

- Detailed description of Busy Indicator 299
 - Busy Tone Disconnect 299
- Hardware requirements for Busy Indicator 299
- Reports for Busy Indicator 300
- Considerations for Busy Indicator 300
- Interactions for Busy Indicator 300

Busy Verification	301
• Detailed description of Busy Verification	301
• Hardware requirements for Busy Verification	303
• Administering Busy Verification	303
Prerequisites for administering Busy Verification	304
Screens for administering Busy Verification	304
Using Busy Verification	304
• Reports for Busy Verification	305
• Considerations for Busy Verification	305
• Interactions for Busy Verification	305
 Call-by-Call Service Selection	 307
• Detailed description of Call-by-Call Service Selection	307
ISDN messages and information elements for usage allocation	308
Usage Allocation Plans	309
Incoming call-handling treatment	310
Call Detail Recording	310
• Hardware requirements for Call-by-Call Service Selection	311
• Administering Call-by-Call Service Selection	311
Prerequisites for administering Call-by-Call Service Selection	312
Screens for administering Call-by-Call Service Selection	312
Setting up a trunk group for CBC	313
Administering route patterns for the CBC trunk group	315
Administering network facilities	316
• Reports for Call-by-Call Service Selection	317
• Considerations for Call-by-Call Service Selection	318
• Interactions for Call-by-Call Service Selection	318
 Call Charge Information	 319
• Detailed description of Call Charge Information	319
Charge Display	320
• Hardware requirements for Call Charge Information	321
• Administering Call Charge Information	321
Prerequisites for administering Call Charge Information	321
Defining CDR to support Call Charge information	322

Specifying the frequency of the call charge displays	323
Translating the text “Call Charge”	324
Assigning a COR for charge displays	325
Screens for administering Call Charge Information	326
Administering the charge display	327
Administering a trunk group for call charge displays	328
Assigning a call charge display button for a user	329
Assigning a call charge display feature button for an attendant	331
Administering AOC for ISDN trunks	332
Administering PPM for non-ISDN trunks	334
Administering PPM for DS1 circuit packs	336
• End-user procedures for Call Charge Information	338
• Reports for Call Charge Information	339
• Considerations for Call Charge Information	339
• Interactions for Call Charge Information	339
Call Coverage	343
• Detailed description of Call Coverage	343
What is a Call Coverage Path?	344
Multiple coverage paths	345
Time-of-Day Coverage	345
Off-Network Call Coverage	346
Call Coverage changeable coverage paths	347
Extended User Administration of Redirected Calls	347
Call coverage criteria	347
VDN in a call coverage path (VICP)	348
Coverage answer groups	349
Announcement in a coverage path	349
Hunt group in a coverage path	349
Subsequent redirection interval	349
Notifying users when the calls are redirected	349
Caller response interval	350
Consult	350
Features that override Call Coverage	350
Conditions that override Call Coverage	351
• Hardware requirements for Call Coverage	352

• Administering Call Coverage	353
Prerequisites for administering Call Coverage	353
Screens for administering Call Coverage	354
Creating a coverage path	355
Prerequisites	355
Assigning a coverage path to a user	358
Prerequisites	358
Assigning a Consult button for a user	360
Defining coverage redirected off-network calls	361
Prerequisites	361
Assigning the telephone numbers for the off-network coverage points	361
Administering the coverage path for redirected off-network calls	362
Assigning time-of-day coverage	363
Prerequisites	363
Creating a coverage answer group	365
Assigning Internal Alerting	366
• Reports for Call Coverage	366
• Considerations for Call Coverage	367
• Interactions for Call Coverage	367
• Troubleshooting Call Coverage	370

Call Detail Recording 373

• Detailed description of Call Detail Recording	373
Monitor call detail records	374
Answer detection	374
Call classification	374
Network Answer Supervision	375
Answer Supervision by Timeout	375
Account Code Dialing	375
Forced Entry of Account Codes	375
Call Splitting	376
Incoming trunk call splitting	376
An example of ITCS and a conference call	377
An example of ITCS and a call transfer on the same server	377
An example of ITCS and a call transfer to the public network	378

Outgoing trunk call splitting	379
An example of OTCS and a conference call on the public network	379
An example of OTCS and a call transfer to a public network	379
ITCS and OTCS and attendant call recording	380
Intrawitch CDR	381
CDR Privacy	382
CDR output	383
CDR record formats	383
Customized CDR call record formats	385
Standard CDR call record formats	385
Call detail record field descriptions	408
• Hardware requirements for Call Detail Recording	420
• Administering Call Detail Recording	421
Prerequisites for administering Call Detail Recording	421
Screens for administering Call Detail Recording	423
Assigning Forced Entry of Account Codes	424
Assigning privacy digits for a user	427
Prerequisites	427
Administering the CDR system parameters	428
Field descriptions for page 2	436
Administering CDR for a trunk group	437
Administering CDR for a data module	440
Identifying the Inter Exchange Carrier	448
Administering CDR for the paging ports	450
Administering the Intra-Switch CDR	450
• End-user procedures for Call Detail Recording	451
• Reports for Call Detail Recording	451
• Considerations for Call Detail Recording	452
• Interactions for Call Detail Recording	452
Call Forwarding	463
• Detailed description of Call Forwarding	463
Call Forwarding All Calls	463
Call Forwarding and FAC	464
Call Forwarding and attendants	464
Call Forward Busy/Don't Answer	464

Call Forwarding Off-Net	465
Call Forwarding Override	465
Notifying users when their calls are redirected	466
Coverage for unanswered forwarded calls	466
Security	466
• Hardware requirements for Call Forwarding	466
• Administering Call Forwarding	466
Prerequisites for administering the Call Forwarding feature	467
Screens for administering Call Forwarding	469
Viewing the user extensions that have the Call Forwarding capabilities active	469
Assigning the Call Forwarding All Calls capability to a user	469
Removing the Call Forwarding All Calls capability for a user	470
Assigning the Call Forward Busy/Don't Answer capability to a user	470
Prerequisites	470
Removing the Call Forward Busy/Don't Answer capability for a user	471
Assigning the Call Forwarding Off Net capability to a user	472
Prerequisites	472
Removing the Call Forwarding Off Net capability for a user	473
Enabling the Call Forwarding Override capability for your system	474
Disabling the Call Forwarding Override capability for your system	474
• End-user procedures for Call Forwarding	475
Changing the Call Forwarding All Calls destination from an internal telephone	475
Changing the Call Forward Busy/Don't Answer destination from an internal telephone	475
Changing the forwarding destination when a user is at an off-network location	475
Changing the Call Forward Busy/Don't Answer destination when a user is at an off-network location	476
• Reports for Call Forwarding	477
• Considerations	477
• Interactions	477

Call Park 481

- Detailed description of Call Park 481
- Hardware requirements for Call Park 481
- Administering Call Park 482
 - Prerequisites for administering the Call Park feature 482
 - Screens for administering Call Park 486
 - Assigning a call park button to a user of a multiple-call appearance telephone 486
- End-user procedures for Call Park 487
 - Using Call Park from a single-line telephone 487
 - Using Call Park from a multiple-call appearance telephone 488
 - Using Call Park from an attendant console 488
 - Retrieving a parked call 488
- Reports for Call Park 488
- Considerations for Call Park 489
- Interactions for Call Park 489

Call Pickup 491

- Detailed Description 491
 - Call Pickup 491
 - Call ringing and status lamp flashing 491
 - Directed Call Pickup 492
 - Extended Group Pickup 492
- Hardware requirements for Call Pickup 493
- Administering Call Pickup 493
 - Screens for administering Call Pickup 493
 - Assigning a call pickup button to a user telephone 494
 - Prerequisites 494
 - Changing a call pickup button for a user telephone 496
 - Removing a call pickup button from a user telephone 496
 - Assigning a user to a call pickup group 496
 - Removing a user from a call pickup group 498
 - Assigning the Directed Call Pickup capability to a user 498
 - Prerequisites 498
 - Removing the Directed Call Pickup capability from a user 501

Assigning a pickup group to an extended pickup group	501
Prerequisites	501
Removing a pickup group from an extended pickup group	504
Assigning extended pickup groups to another extended pickup group	504
Prerequisites	504
Removing an extended pickup group from an extended pickup group	507
• End-user procedures for Call Pickup	508
• Reports for Call Pickup	508
• Considerations for Call Pickup	508
• Interactions for Call Pickup	508
Call Waiting Termination	511
• Detailed description of Call Waiting Termination	511
• Hardware requirements for Call Waiting Termination	511
• Administering Call Waiting Termination	512
Prerequisites for administering Call Waiting Termination	512
Screens for administering Call Waiting Termination	513
Assigning Call Waiting Termination	513
• Reports for Call Waiting Termination	515
• Considerations for Call Waiting Termination	515
• Interactions for Call Waiting Termination	515
Caller ID	517
• Detailed description of Caller ID	517
Caller ID (ICLID) on analog trunks	517
Caller ID (ICLID) on digital trunks	517
• Hardware requirements for Caller ID	518
• Administering Caller ID	518
Prerequisites for administering Caller ID	518
Screens for administering Caller ID	518
Displaying ICLID Information	519
• Reports for Caller ID	520
• Considerations for Caller ID	521
• Interactions for Caller ID	521

Centralized Attendant Service 523

- Detailed description of Centralized Attendant Service 523
 - Branch-generated call identification tones 524
- Hardware requirements for Centralized Attendant Service 525
- Administering Centralized Attendant Service 525
 - Prerequisites for administering Centralized Attendant Service 525
 - Screens for administering Centralized Attendant Service 525
- Reports for Centralized Attendant Service 526
- Considerations for Centralized Attendant Service 526
- Interactions for Centralized Attendant Service 527

Class of Restriction 531

- Detailed description of Class of Restriction 531
 - Strategy for assigning CORs 532
 - Types of Restrictions 532
 - Calling party restrictions 532
 - Called party restrictions 533
 - Fully restricted service 535
 - COR-to-COR restrictions 535
- Hardware requirements for Class of Restriction 535
- Administering Class of Restriction 535
 - Prerequisites for administering Class of Restriction 535
 - Screens for administering Class of Restriction 536
 - Displaying administered CORs 536
 - Setting up a COR 537
 - Allowing users to change their own COR 540
 - Assigning a Feature Access Code (FAC) 541
 - Assigning a password 542
- End-user procedures for Class of Restriction 542
- Reports for Class of Restriction 542
- Considerations for Class of Restriction 543
- Interactions for Class of Restriction 543

Class of Service	547
• Detailed description of Class of Service	547
• Hardware requirements for Class of Service	547
• Administering Class of Service	547
Prerequisites for administering Class of Service	548
Screens for administering Class of Service	548
Defining COS for your system	548
Descriptions of the COS features	549
Assigning a COS	551
• Reports for Class of Service	551
• Considerations for Class of Service	552
• Interactions for Class of Service	552
 Co-resident DEFINITY LAN Gateway	 555
• Detailed description of Co-resident DEFINITY LAN Gateway	555
How Co-resident DLG works	555
• Hardware requirements for Co-resident DEFINITY LAN Gateway	556
CallVisor ASAI CTI link	556
Co-resident DLG	556
• Administering Co-resident DEFINITY LAN Gateway	556
Prerequisites for administering Co-resident DEFINITY LAN Gateway	557
TN799 C-LAN circuit pack as the Ethernet Interface	557
TN2314 circuit pack as the Ethernet Interface	557
Screens for administering Co-resident DEFINITY LAN Gateway	558
• Reports for Co-resident DEFINITY LAN Gateway	559
• Considerations for Co-resident DEFINITY LAN Gateway	559
• Interactions for Co-resident DEFINITY LAN Gateway	559
 Conference	 561
• Detailed description of Conference	561
Conference and DCP, hybrid, IP, wireless, and ISDN-BRI telephones	562
Conference/Transfer Toggle/Swap	562
No Dial Tone Conferencing	562

No Hold Conference	562
Select Line Appearance Conferencing	563
Selective Conference Party Display, Drop, and Mute	563
• Hardware requirements for Conference	563
• Administering Conference	564
Prerequisites for administering the Conference feature	564
Screens for administering Conference	566
Assigning the toggle-mute feature button	567
Assigning the conf-dsp, fe-mute, and no-hold-conf feature buttons	568
Prerequisites	568
Assigning the conf-dsp, fe-mute, and no-hold-conf feature buttons to a user	569
Assigning the conf-dsp and fe-mute feature buttons to an attendant	570
• End-user procedures for Conference	571
Displaying the participants on a conference call	571
• Reports for Conference	572
• Considerations for Conference	572
• Interactions for Conference	573

Data Call Setup 575

• Detailed description of Data Call Setup	575
• Hardware requirements for Data Call Setup	578
• Administering Data Call Setup	578
Prerequisites	578
Screens for administering Data Call Setup	579
Defining a data module	580
Specifying the port location	590
Assigning the data extension feature button	590
• End-user procedures for Data Call Setup	591
Setting up and disconnecting data calls from a DCP data terminal	592
Setting up data calls from a DCP telephone	592
Setting up and disconnecting data calls from an ISDN-BRI data terminal	593
Setting up data calls from an ISDN-BRI telephone	593
• Reports for Data Call Setup	593

• Considerations for Data Call Setup	594
• Interactions for Data Call Setup	594
Default Dialing	597
• Detailed description of Default Dialing	597
• Hardware requirements for Default Dialing	597
• Administering Default Dialing	597
Screens for administering Default Dialing	598
• Reports for Default Dialing	598
• Considerations for Default Dialing	598
• Interactions for Default Dialing	598
Demand Print	599
• Detailed description of Demand Print	599
• Hardware requirements for Demand Print	599
• Administering Demand Print	599
Screens for administering Demand Print	599
• Reports for Demand Print	600
• Considerations for Demand Print	600
• Interactions for Demand Print	600
Dial Access to Attendant	601
• Detailed description of Dial Access to Attendant	601
• Hardware requirements for Dial Access to Attendant	601
• Administering Dial Access to Attendant	601
Prerequisites for administering Dial Access to Attendant	601
Screens for administering Dial Access to Attendant	602
Changing the attendant access code	602
• Reports for Dial Access to Attendant	603
• Considerations for Dial Access to Attendant	603
• Interactions for Dial Access to Attendant	603

Dial Plan	605
• Detailed Description of Dial Plan	605
Dial Plan Analysis Table	605
Dial Plan Parameters	606
Multi-Location Dial Plan	607
Other options for Dial Plan	608
Uniform Dial Plan (UDP)	608
• Hardware Requirements for Dial Plan	608
• Administering Dial Plan	609
Prerequisites for administering Dial Plan	609
Screens for administering Dial Plan	609
Displaying your dial plan	610
Modifying your dial plan	610
Adding extension ranges	611
Adding Feature Access Codes	611
• Reports for Dial Plan	611
• Considerations for Dial Plan	612
• Interactions for Dial Plan	612
 Distinctive Ringing	 615
• Detailed description of Distinctive Ringing	615
• Hardware requirements for Distinctive Ringing	615
• Administering Distinctive Ringing	615
Prerequisites for administering Distinctive Ringing	616
Screens for administering Distinctive Ringing	616
Defining Distinctive Ringing	616
• Reports for Distinctive Ringing	618
• Considerations for Distinctive Ringing	618
• Interactions for Distinctive Ringing	618
 Enhanced 911	 619
• Detailed description of Enhanced 911	619
Configurations with media gateways in different locations	620
Location Specific Routing	621

E911 for wired IP telephones	622
Emergency Extension Forwarding	623
Call Forwarding of dropped emergency calls	623
Crisis Alert	624
Cancelling an alert	624
• Hardware requirements for Enhanced 911	625
• Administering Enhanced 911	625
Prerequisites for administering Enhanced 911	626
Screens for administering Enhanced 911	626
Setting up Crisis Alert to an attendant or a display telephone	627
Setting up the emergency number	627
Setting up the attendant console to receive emergency notification	629
Setting up digital telephones to receive emergency notification	629
Setting which users must acknowledge the emergency alert	630
Setting up Crisis Alert to notify a digital pager	631
Prerequisites	631
Setting up emergency extension forwarding	633
Setting up CAMA numbering	634
• Reports for Enhanced 911	635
• Considerations for Enhanced 911	636
• Interactions for Enhanced 911	637
Extended User Administration of Redirected Calls	639
• Detailed description of Extended User Administration of Redirected Calls	639
Disabling the telecommuting access extension	639
Extended User Administration of Redirected Calls and DCS	640
Extended User Administration of Redirected Calls and Class of Service	640
Extended User Administration of Redirected Calls and COR	640
Extended User Administration of Redirected Calls from an off-site telephone	640
• Hardware requirements for Extended User Administration of Redirected Calls	641

• Administering Extended User Administration of Redirected Calls	641
Prerequisites for administering Extended User Administration of Redirected Calls	642
Screens for administering Extended User Administration of Redirected Calls	645
Assigning a station security code to a user	646
• End-user procedures for Extended User Administration of Redirected Calls	647
Changing Call Coverage or activating Call Forwarding from an on-site telephone that is not assigned to the user	648
Deactivating Call Forwarding from an on-site telephone that is not assigned to the user	648
• Reports for Extended User Administration of Redirected Calls	649
• Considerations for Extended User Administration of Redirected Calls	649
• Interactions for Extended User Administration of Redirected Calls	649

Facility and Non-Facility Associated Signaling **651**

• Detailed description of Facility and Non-Facility Associated Signaling	651
D-channel backup with NFAS	651
D-channel backup activation	652
• Hardware requirements for Facility and Non-Facility Associated Signaling	653
• Administering Facility and Non-Facility Associated Signaling	653
Prerequisites for administering Facility and Non-Facility Associated Signaling	653
Screens for administering Facility and Non-Facility Associated Signaling	653
Reviewing the guidelines for FAS and NFAS	654
Implementing FAS and NFAS	655
Administering the DS1 Circuit Pack screen	655
Administering the Interface Links and Processor Channels screens	655
Administering the ISDN-PRI Trunk Group, Signaling Group, and Trunk Group Members screens	656
• Reports for Facility and Non-Facility Associated Signaling	658

- Considerations for Facility and Non-Facility Associated Signaling 658
- Interactions for Facility and Non-Facility Associated Signaling 658

Facility Restriction Levels 659

- Detailed description of Facility Restriction Levels 659
 - AAR and ARS calls 659
 - Originators of AAR and ARS calls 659
 - Call termination points for AAR and ARS calls 660
 - AFRL 660
 - Alt-fri feature button 661
 - Authorization codes 661
- Hardware requirements for Facility Restriction Levels 661
- Administering Facility Restriction Levels 661
 - Screens for administering Facility Restriction Levels 662
- End-user procedures for Facility Restriction Levels 663
- Reports for Facility Restriction Levels 663
- Considerations for Facility Restriction Levels 663
- Interactions for Facility Restriction Levels 664

Feature Access Codes 665

- Detailed description of Feature Access Codes 665
- Hardware requirements for Feature Access Codes 666
- Administering Feature Access Codes 666
 - Prerequisites for administering Feature Access Codes 666
 - Screens for administering Feature Access Codes 666
 - Assigning Feature Access Codes 666
 - Changing or deleting Feature Access Codes 667
- Reports for Feature Access Codes 668
- Considerations for Feature Access Codes 668
- Interactions for Feature Access Codes 669
- Troubleshooting Feature Access Codes 669

Group Paging 671

- Detailed Description of Group Paging 671
 - Restrictions 671
 - Controlling access to paging groups 671
- Hardware requirements for Group Paging 671
- Administering Group Paging 672
 - Prerequisites for administering Group Paging 672
 - Screens for administering Group Paging 672
 - Creating a Paging Group 672
 - Changing a Paging Group 673
 - Viewing All Paging Groups 674
- Reports for Group Paging 674
- Considerations for Group Paging 674
- Interactions for Group Paging 674
- Troubleshooting Group Paging 676

Hold 677

- Detailed description of Hold 677
 - Soft Hold 677
 - Hard Hold 677
 - Automatic Hold 677
- Hardware requirements for Hold 678
- Administering Hold 678
 - Prerequisites for administering Hold 678
 - Screens for administering Hold 678
 - Enabling Automatic Hold 679
 - Assigning an FAC for CAS remote hold and answer 679
- Reports for Hold 680
- Considerations for Hold 681
- Interactions for Hold 681

Hot Line Service 683

- Detailed description of Hot Line Service 683
- Hardware requirements for Hot Line Service 684
- Administering Hot Line Service 684
 - Screens for administering Hot Line Service 684

- Reports for Hot Line Service 685
- Considerations for Hot Line Service 685
- Interactions for Hot Line Service 685

Hunt Groups 687

- Detailed description of Hunt Groups 688
 - Announcements for hunt groups 688
 - Delay announcement intervals for hunt groups 688
 - Analog, aux-trunk, or integrated delay announcements for hunt groups 689
 - Call Coverage for hunt groups 689
 - Call Distribution methods for hunt group types 689
 - Hunt group extension unavailability 691
 - Hunt Group Busy option 691
 - Using Send All Calls with hunt groups 692
 - Using Call Forwarding All Calls with hunt groups 692
 - Queues for hunt groups 692
 - TTY hunt groups 692
- Hardware requirements for Hunt Groups 693
- Administering Hunt Groups 693
 - Prerequisites for administering Hunt Groups 693
 - Screens for administering Hunt Groups 694
 - Setting up hunt groups 694
 - Changing hunt groups 696
 - Setting up queues for hunt groups 696
 - Adding hunt group announcements 697
 - Setting up night service for hunt groups 698
- Reports for Hunt Groups 699
- Considerations for Hunt Groups 699
- Interactions for Hunt Groups 700

Individual Attendant Access 703

- Detailed description of Individual Attendant Access 703
- Hardware requirements for Individual Attendant Access 703
- Administering Individual Attendant Access 703
 - Prerequisites for administering Individual Attendant Access 703
 - Screens for administering Individual Attendant Access 704

Assigning an extension to an attendant console	704
• Reports for Attendant Individual Attendant Access	705
• Considerations for Individual Attendant Access	705
• Interactions for Individual Attendant Access	705
Inter-PBX Attendant Service	707
• Detailed description of Inter-PBX Attendant Service	707
• Hardware requirements for Inter-PBX Attendant Service	707
• Administering Inter-PBX Attendant Service	707
Prerequisites for administering Inter-PBX Attendant Service	708
Screens for administering Inter-PBX Attendant Service	708
Enabling Inter-PBX Attendant Service	708
• Reports for Inter-PBX Attendant Service	709
• Considerations for Inter-PBX Attendant Service	710
• Interactions for Inter-PBX Attendant Service	710
Intercom	711
• Detailed description of Intercom	711
Intercom groups	711
Telephones	712
• Hardware requirements for Intercom	712
• Administering Intercom	712
Screens for administering Intercom	712
• Reports for Intercom	712
• Considerations for Intercom	713
• Interactions for Intercom	713
Internal Automatic Answer	715
• Detailed description of Internal Automatic Answer	715
Feature operations	716
• Hardware requirements for Internal Automatic Answer	716
• Administering Internal Automatic Answer	716
Screens for administering Internal Automatic Answer	716
• Reports for Internal Automatic Answer	716

- Considerations for Internal Automatic Answer 717
- Interactions for Internal Automatic Answer 717

ISDN Service 721

- Detailed description of ISDN Service 721
 - Transmission rate and protocols 722
 - AT&T Switched Network protocol 723
 - Private network services 724
 - ETN services 724
 - DCS services 725
 - QSIG services 725
 - Wideband Switching (ISDN-PRI only) 725
 - Call-by-Call Service Selection 725
 - Access to Software Defined Data Network 725
 - Access to Switched Digital International 725
 - National ISDN-2 services 725
 - Calling Line Identification 726
 - Non-Facility Associated Signaling (ISDN-PRI only) 726
 - D-Channel Backup (ISDN-PRI only) 726
 - Wideband Switching (ISDN-PRI only) 726
 - Call-by-Call Service Selection 726
 - ISDN interworking 726
 - Call Identification Display 727
 - Displays for redirected calls 729
 - Displays for conference calls 729
 - Displays for calls to hunt groups 729
 - Displays for calls to Terminating Extension Groups (TEG) 729
 - Caller Information Forwarding 729
 - Facility Restriction Level (FRL) and Traveling Class Mark (TCM) 730
 - Information Indicator Digits (II-digits) 730
 - Malicious Call Trace (MCT) 730
 - Multiple Subscriber Number (MSN) - Limited 730
 - Overlap Sending 730
 - Interworking between TGU/TGE trunks and ISDN (Italy) 730
- Hardware requirements for ISDN Service 730
- Administering ISDN Service 731
 - Screens for administering ISDN Service 731
- Reports for ISDN Service 731

• Considerations for ISDN Service	731
• Interactions for ISDN Service	732
Last Number Dialed	735
• Detailed description of Last Number Dialed	735
• Hardware requirements for Last Number Dialed	735
• Administering Last Number Dialed	735
Screens for administering Last Number Dialed	735
• Reports for Last Number Dialed	736
• Considerations for Last Number Dialed	736
• Interactions for Last Number Dialed	736
Leave Word Calling	737
• Detailed description of Leave Word Calling	737
Voice synthesis	738
• Hardware requirements for Leave Word Calling	738
• Administering Leave Word Calling	738
• Considerations for Leave Word Calling	739
• Interactions for Leave Word Calling	739
Line Lockout	741
• Detailed description of Line Lockout	741
• Hardware requirements for Line Lockout	741
• Administering Line Lockout	742
Screens for administering Line Lockout	742
• Reports for Line Lockout	742
• Considerations for Line Lockout	743
• Interactions for Line Lockout	743
Listed Directory Number	745
• Detailed description of Listed Directory Number	745
Incoming DID trunk calls to the attendant group	745
Incoming FX and CO trunk calls to the attendant group	745
• Hardware requirements for Listed Directory Number	745

• Administering Listed Directory Number	746
Prerequisites for administering Listed Directory Number	746
Screens for administering Listed Directory Number	746
Assigning listed directory numbers	746
Assigning an incoming destination to a trunk	747
• Reports for Listed Directory Number	749
• Considerations for Listed Directory Number	749
• Interactions for Listed Directory Number	749

Loss Plans 751

• Detailed description of Loss Plans	751
• Hardware requirements for Loss Plans	751
• Administering Loss Plans	752
Prerequisites for administering Loss Plans	752
Screens for administering Loss Plans	752
• Reports for Loss Plans	752
• Considerations for Loss Plans	753
• Interactions for Loss Plans	753

Loudspeaker Paging 755

• Detailed description of Loudspeaker Paging	755
Types of Paging	755
Deluxe Paging	755
Chime paging	756
Auxiliary paging systems	756
Restrictions on loudspeaker paging	756
• Hardware requirements for Loudspeaker Paging	757
• Administering Loudspeaker Paging	757
Prerequisites for administering Loudspeaker Paging	757
Screens for administering Loudspeaker Paging	758
Setting up Voice Paging over loudspeakers	758
Setting up Chime Paging over Loudspeakers	760
Assigning a chime page code to an individual extension	761
• Reports for Loudspeaker Paging	762
• Considerations for Loudspeaker Paging	762

- Interactions for Loudspeaker Paging 762
- Troubleshooting Loudspeaker Paging 764

Malicious Call Trace 765

- Detailed description of Malicious Call Trace 765
 - Activating MCT 765
 - Deactivating MCT 766
 - Using the MCT voice recorder 766
 - Controlling MCT 766
 - Administering MCT for ISDN notification 767
- Hardware requirements for Malicious Call Trace 767
- Administering Malicious Call Trace 767
 - Screens for administering Malicious Call Trace 767
- Reports for Malicious Call Trace 768
- Considerations for Malicious Call Trace 768
- Interactions for Malicious Call Trace 769

Manual Message Waiting 771

- Detailed description of Manual Message Waiting 771
- Hardware requirements for Manual Message Waiting 771
- Administering Manual Message Waiting 771
 - Prerequisites for administering Manual Message Waiting 772
 - Screens for administering Manual Message Waiting 772
 - Assigning the Manual Message Waiting feature button 772
- Reports for Manual Message Waiting 773
- Considerations for Manual Message Waiting 773
- Interactions for Manual Message Waiting 773

Manual Signaling 775

- Detailed description of Manual Signaling 775
- Hardware requirements for Manual Signaling 775
- Administering Manual Signaling 775
 - Screens for administering Manual Signaling 775
- End-user procedures for Manual Signaling 776
- Reports for Manual Signaling 776

• Considerations for Manual Signaling	776
• Interactions for Manual Signaling	776
Meet-me Conference	777
• Detailed Description of Meet-me Conference	777
• Hardware requirements for Meet-me Conference	777
• Administering Meet-me Conference	777
Prerequisites for administering Meet-me Conference	778
Screens for administering Meet-me Conference	778
Creating or changing a Meet-me Conference vector	778
Options for creating vector steps	780
How the vector processes a call	781
Creating a Meet-me Conference VDN	782
• End-user procedures for Meet-me Conference	784
Accessing a Meet-me Conference as an attendee	784
Changing a Meet-me Conference access code	784
Using Selective Conference Party Display, Drop, and Mute	785
• Reports for Meet-me Conference	786
• Considerations for Meet-me Conference	786
• Interactions for Meet-me Conference	788
• Troubleshooting Meet-me Conference	789
Modem Pooling	791
• Detailed description of Modem Pooling	791
• Hardware requirements for Modem Pooling	792
• Administering Modem Pooling	792
Screens for administering Modem Pooling	792
• Reports for Modem Pooling	792
• Considerations for Modem Pooling	793
• Interactions for Modem Pooling	793
Multi-Location Dial Plan	795
• Detailed description of Multi-Location Dial Plan	796
Location prefix	796
Feature Access Codes	797
Announcements	797

Invalid announcements	798
Maintenance	798
• Hardware requirements for Multi-Location Dial Plan	798
• Administering Multi-Location Dial Plan	799
Prerequisites for administering Multi-Location Dial Plan	799
Screens for administering Multi-Location Dial Plan	799
Changing extensions	800
Function	801
Exceptions	801
Audits	801
Prepending numbers to the dialed string	802
Setting up announcements	803
Setting up a local centralized answering point	803
Setting up multiple feature access codes for attendants	804
Setting up multiple feature access codes for ARS	804
• Reports for Multi-Location Dial Plan	804
• Considerations for Multi-Location Dial Plan	804
• Interactions for Multi-Location Dial Plan	805
Multifrequency Signaling	807
• Detailed description of Multifrequency Signaling	807
MFE	808
MF Shuttle	808
R2-MFC	808
Guidelines for administering Multifrequency Signaling	808
• Hardware requirements for Multifrequency Signaling	809
• Administering Multifrequency Signaling	809
Screens for administering Multifrequency Signaling	810
• Reports for Multifrequency Signaling	810
• Considerations for Multifrequency Signaling	810
• Interactions for Multifrequency Signaling	811
Multiple Level Precedence and Preemption	815
• Detailed description of Multiple Level Precedence and Preemption	815
Precedence Calling	816
Precedence Calling diversion scenarios	817

How service domains influence preemption and precedence	819
Announcements for Precedence Calling	824
Precedence Call Waiting	825
Precedence Routing	825
Dual Homing	825
End Office Access Line Hunting	826
Preemption	826
Line Load Control	826
Worldwide Numbering and Dialing Plan	828
• Hardware requirements for Multiple Level Precedence and Preemption	830
• Administering Multiple Level Precedence and Preemption	831
Prerequisites for administering Multiple Level Precedence and Preemption	832
Screens for administering Multiple Level Precedence and Preemption	832
Precedence Calling	834
Assigning an MLPP feature access code	834
Assigning Precedence Calling system parameters	834
Assigning a maximum precedence level	836
Assigning trunks	836
Assigning a hot line number	838
Assigning attendant queue priorities	839
Announcements for Precedence Calling	840
Adding extensions	840
Assigning announcement types	841
Recording announcements	841
Deleting announcements	842
Saving announcements	843
Precedence Call Waiting	843
Enabling Precedence Call Waiting for a telephone	843
Setting the Precedence Call timeout	843
Assigning feature access codes	844
Precedence Routing	844
Assigning digit analysis	845
Assigning route patterns	847
Assigning digit conversion	847
Dual Homing	848

End Office Access Line Hunting	848
Preemption	849
Assigning Preemption to a COR	849
Assigning trunks for Preemption	849
Setting the Precedence Call timeout	849
Line Load Control	849
Assigning the LLC level for the system	849
Assigning the LLC to a COR	849
Worldwide Numbering and Dialing Plan	850
Assigning WNDP system parameters	850
Assigning WNDP feature access codes	851
Assigning a Hot Line number	851
Assigning attendant queue priorities	852
• Reports for Multiple Level Precedence and Preemption	852
• Considerations for Multiple Level Precedence and Preemption	852
Precedence Calling	852
Announcements for Precedence Calling	853
Precedence Call Waiting	853
Precedence Routing	853
Dual Homing	853
End Office Access Line Hunting	853
Preemption	854
Line Load Control	854
Worldwide Numbering and Dialing Plan (WNDP)	854
• Interactions for Multiple Level Precedence and Preemption	854
Precedence Calling	854
Announcements for Precedence Calling	857
Precedence Call Waiting	857
Precedence Routing	858
Dual Homing	858
End Office Access Line Hunting	859
Preemption	859
Line Load Control	861
Worldwide Numbering and Dialing Plan (WNDP)	862

Music-on-Hold	863
• Detailed description of Music-on-Hold	863
• Hardware requirements for Music-on-Hold	863
• Administering Music-on-Hold	864
Prerequisites for administering Music-on-Hold	864
Screens for administering Music-on-Hold	864
Assigning music tones, music ports, and music for transferred trunks	865
Defining a Class of Restriction for Music-on-Hold	866
Connecting a music source to the server	867
Assigning a source of music to a port	867
• Reports for Music-on-Hold	868
• Considerations for Music-on-Hold	869
• Interactions for Music-on-Hold	869
Night Service	871
• Detailed description of Night Service	871
Hunt Group Night Service	871
Night Console Service	871
Night Station Service	871
Trunk Answer from Any Station (TAAS)	872
Trunk Group Night Service	872
• Hardware requirements for Night Service	873
• Administering Night Service	873
Screens for administering Night Service	873
Setting up night station service to voice mail	874
Setting up Night Console Service	876
Setting up Night Station Service	877
Setting up Trunk Answer from Any Station	879
Setting up external alerting	879
Setting up external alerting Night Service	880
Setting up Night Service for trunk groups	882
Setting up Night Service for hunt groups	882
• Reports for Night Service	883

• Considerations for Night Service	883
Considerations for Hunt Group Night Service	883
Considerations for Night Console Service	884
Considerations for Night Station Service	884
Considerations for TAAS	884
Considerations for Trunk Group Night Service	884
• Interactions for Night Service	885
Interactions for Hunt Group Night Service	885
Interactions for Night Console Service	886
Interactions for Night Station Service	886
Interactions for TAAS	887
Interactions for Trunk Group Night Service	887

Off-Premises Station 889

• Detailed description of Off-Premises Station	889
• Hardware requirements for Off-Premises Station	889
• Administering Off-Premises Station	890
Prerequisites for administering Off-Premises Station	890
Screens for administering Off-Premises Station	890
Activating Off-Premises Station for a user	890
• Reports for Off-Premises Station	891
• Considerations for Off-Premises Station	892
• Interactions for Off-Premises Station	892

Personal Station Access 893

• Detailed description of Personal Station Access	893
Telecommuting	893
Invalid attempts to use PSA	893
Preferences and permissions	893
Button mapping	894
Unanswered calls	894
Dissociated telephones	894
Security	894
• Hardware requirements for Personal Station Access	895
• Administering Personal Station Access	895
Screens for administering Personal Station Access	895

• End-user procedures for Personal Station Access	896
Using the PSA associate command	896
Interrupting the PSA associate command sequence	898
Using the PSA dissociate command	898
• Reports for Personal Station Access	898
• Considerations for Personal Station Access	898
• Interactions for Personal Station Access	899

Personalized Ringing 901

• Detailed description of Personalized Ringing	901
Ringing patterns	901
Power failures	901
• Hardware requirements for Personalized Ringing	901
• Administering Personalized Ringing	902
Prerequisites for administering Personalized Ringing	902
Screens for administering Personalized Ringing	902
Assigning Personalized Ringing to a user telephone	902
• Reports for Personalized Ringing	904
• Considerations for Personalized Ringing	904
• Interactions for Personalized Ringing	904

Posted Messages 905

• Detailed description of Posted Messages	905
Language options	905
Messages	906
Fixed messages	906
Custom messages	907
QSIG support	907
• Hardware requirements for Posted Messages	908
• Administering Posted Messages	908
Prerequisites for administering Posted Messages	908
Screens for administering Posted Messages	909
Defining a feature access code (FAC)	910
Requiring a security code	911
Setting up a security code for the telephone of a user	911
Activating the Posted Messages security code	912
Activating QSIG to send custom messages	913

Translating Posted Messages	914
Translating fixed messages to a user-defined language	914
Creating custom messages	915
Translating telephone feature buttons and labels	916
Translating the Posted Messages feature display to a user-defined language	917
Translating the Posted Messages softkey button label to a user-defined language	917
Translating the Posted Messages button label to a user-defined language	918
• End-user procedures for Posted Messages	919
Activating the Posted Messages feature	920
Using a feature access code (FAC)	920
Examples	920
Using the Posted Messages feature button	921
Deactivating the Posted Messages feature	921
Using a feature access code (FAC)	921
Examples	922
Using the Posted Messages feature button	922
• Reports for Posted Messages	922
• Considerations for Posted Messages	923
• Interactions for Posted Messages	924
Priority Calling	927
• Detailed description of Priority Calling	927
• Hardware requirements for Priority Calling	927
• Administering Priority Calling	927
Prerequisites for administering Priority Calling	928
Screens for administering Priority Calling	929
Assigning a priority feature button to an attendant console	930
Assigning a priority feature button to a telephone	931
• End-user procedures for Priority Calling	931
• Reports for Priority Calling	932
• Considerations for Priority Calling	932
• Interactions for Priority Calling	932

Privacy	935
• Detailed description of Privacy	935
Data Privacy	935
Data Restriction	935
Privacy - Automatic Exclusion	935
Privacy - Manual Exclusion	936
• Hardware requirements for Privacy	936
• Administering Privacy	936
Prerequisites for administering Privacy	936
Screens for administering Privacy	937
Administering Privacy for a user	937
Activating Data Restriction for a trunk group	939
• End-user procedures for Privacy	940
• Reports for Privacy	940
• Considerations for Privacy	940
• Interactions for Privacy	941
 Public Network Call Priority	 943
• Detailed description of Public Network Call Priority	943
China	943
Forced disconnect	943
Mode-of-release control	943
Russia	944
Intrusion	944
Re-ring	944
Spain	945
Call retention	945
Re-ring	945
• Hardware requirements for Public Network Call Priority	945
• Administering Public Network Call Priority	945
Screens for administering Public Network Call Priority	945
Screens for China	945
Screens for Russia	946
Screens for Spain	946
• Reports for Public Network Call Priority	946
• Considerations for Public Network Call Priority	946

• Interactions for Public Network Call Priority	946
China	947
Forced disconnect	947
Mode-of-release control	947
Re-ring	948
Russia	948

Recorded Telephone Dictation Access 951

• Detailed description of Recorded Telephone Dictation Access	951
• Hardware requirements for Recorded Telephone Dictation Access	951
• Reports for Recorded Telephone Dictation Access	951
• Considerations for Recorded Telephone Dictation Access	952
• Interactions for Recorded Telephone Dictation Access	952

Remote Access 953

• Detailed description of Remote Access	953
Night Service	954
Security	954
Status remote-access	954
Barrier codes	955
Authorization codes	955
Alternate Facility Restriction Levels (AFRLs)	956
Class of Restriction	956
Logoff Notification	956
• Hardware requirements for Remote Access	956
• Administering Remote Access	956
Prerequisites for administering Remote Access	957
Screens for administering Remote Access	957
Administering the Remote Access screen	957
Enabling Remote Access	957
Disabling Remote Access	959
Assigning authorization codes	960
Prerequisites	960
Administering Remote Access for Night Service	963
• End-user procedures for Remote Access	963
• Reports for Remote Access	964

• Considerations for Remote Access	964
• Interactions for Remote Access	964
Restriction - Controlled	965
• Detailed description of Restriction - Controlled	965
• Hardware requirements for Restriction - Controlled	965
• Administering Restriction - Controlled	966
Screens for administering Restriction - Controlled	966
• End-user procedures for Restriction - Controlled	966
Activating Restriction - Controlled	967
• Reports for Restriction - Controlled	967
• Considerations for Restriction - Controlled	967
• Interactions for Restriction - Controlled	967
Ringing - Abbreviated and Delayed	969
• Detailed description of Ringing - Abbreviated and Delayed	969
• Hardware requirements for Ringing - Abbreviated and Delayed	970
• Administering Ringing - Abbreviated and Delayed	970
Prerequisites for administering Ringing - Abbreviated and Delayed	970
Screens for administering Ringing - Abbreviated and Delayed	971
Assigning ring control to a user	972
Assigning an abbreviated ringing button to a user	973
• Reports for Ringing - Abbreviated and Delayed	973
• Considerations for Ringing - Abbreviated and Delayed	974
• Interactions for Ringing - Abbreviated and Delayed	974
Security Violation Notification	977
• Detailed description of Security Violation Notification	977
Security violation thresholds and notification	977
Sequence of events	978
Reporting	978
SVN - halt buttons	978
SVN Referral Call With Announcement	979

Dealing with Security Violations	979
Disabling a login ID	979
Enabling a login ID	979
Enabling remote access	979
Disabling remote access	979
• Hardware requirements for Security Violation Notification	980
• Administering Security Violation Notification	980
Prerequisites for administering Security Violation Notification	980
Screens for administering Security Violation Notification	980
Setting up Security Violation Notification	981
• Reports for Security Violation Notification	983
• Considerations for Security Violation Notification	984
• Interactions for Security Violation Notification	984
Separation of Bearer and Signaling	985
• Detailed description of Separation of Bearer and Signaling	985
• Hardware requirements for Separation of Bearer and Signaling	985
• Administering Separation of Bearer and Signaling	985
Screens for administering Separation of Bearer and Signaling	986
• Reports for Separation of Bearer and Signaling	986
• Considerations for Separation of Bearer and Signaling	986
• Interactions for Separation of Bearer and Signaling	987
Overview	987
Understanding potential interactions	988
General system features	989
Attendant	990
Attendant features that do not work with SBS	990
Attendant features that work with SBS	990
Adjunct Switch Applications Interface (ASAI)	990
Intuity AUDIX and Octel voice mail adjuncts	991
Call Center	991
Automatic Call Distribution (ACD)	991
Best Service Routing	992
Vectors	992

Networking Related Interactions	992
Networking features or capabilities that do not work	992
Networking features or capabilities that work	993
Service Observing	997
• Detailed description of Service Observing	997
Listen and talk modes	998
Restrictions	998
Telephone displays	998
Trunk calls	998
Warning and conference tones	998
• Hardware requirements for Service Observing	999
• Administering Service Observing	999
Screens for administering Service Observing	999
• End-user procedures for Service Observing	1000
• Reports for Service Observing	1001
• Considerations for Service Observing	1001
• Interactions for Service Observing	1001
Station Hunting	1005
• Detailed description of Station Hunting	1005
Station Hunting and Call Coverage	1006
Removing a station from a Station Hunting chain	1006
Duplicating a station in a Station Hunt chain	1006
• Hardware requirements for Station Hunting	1006
• Administering Station Hunting	1007
Prerequisites for administering Station Hunting	1007
Screens for administering Station Hunting	1007
Assigning station hunting after coverage	1007
Assigning a hunt-to station to an extension	1008
Administering Station Hunting before Coverage	1009
• Reports for Station Hunting	1010
• Considerations for Station Hunting	1010
• Interactions for Station Hunting	1011

Station Lock	1015
• Detailed description of Station Lock	1015
• Hardware requirements for Station Lock	1015
• Administering Station Lock	1016
Screens for administering Station Lock	1016
• End-user procedures for Station Lock	1016
• Reports for Station Lock	1017
• Considerations for Station Lock	1017
• Interactions for Station Lock	1017
 Station Security Code	 1019
• Detailed description of Station Security Code	1019
• Hardware requirements for Station Security Code	1019
• Administering Station Security Code	1019
Screens for administering Station Security Code	1019
Creating a station security code	1020
• Reports for Station Security Code	1021
• Considerations for Station Security Code	1021
• Interactions for Station Security Code	1021
 Supporting TTY Callers	 1023
• Detailed description of Supporting TTY Callers	1023
Setting up announcements for TTY callers	1024
Setting up hunt groups for TTY callers	1024
Handling TTY calls with vectors	1025
• Hardware requirements for Supporting TTY Callers	1025
• Administering Supporting TTY Callers	1026
Screens for administering Supporting TTY Callers	1026
• Reports for Supporting TTY Callers	1026
• Considerations for Supporting TTY Callers	1026
• Interactions for Supporting TTY Callers	1026
 Telephone Display	 1027
• Detailed description of Telephone Display	1027
Button display modes	1028
Call-related information display	1029

Message retrieval	1031
Feature information displays	1031
Enhanced Telephone Display	1031
Mapping enhanced display characters	1050
US English to Russian characters	1050
US English to Japanese characters	1051
US English to European characters	1053
US English to Ukrainian characters	1055
• Hardware requirements for Telephone Display	1055
• Administering Telephone Display	1055
Screens for administering Telephone Display	1056
• Reports for Telephone Display	1056
• Considerations for Telephone Display	1056
• Interactions for Telephone Display	1056
Tenant Partitioning	1059
• Detailed description of Tenant Partitioning	1059
Partitioning tenants	1060
Tenant Partitioning examples	1061
Multiple Music-on-Hold	1062
• Hardware requirements for Tenant Partitioning	1063
• Administering Tenant Partitioning	1063
Prerequisites for administering Tenant Partitioning	1064
Screens for administering Tenant Partitioning	1064
Defining a tenant partition	1065
Assigning a tenant partition number to an access telephone	1067
Assigning a tenant partition number to an agent login ID	1068
Assigning a tenant partition number to an announcement	1069
Assigning a tenant partition number to an attendant	1069
Assigning a tenant partition number to a data module	1070
Assigning a tenant partition number to a hunt group	1071
Assigning a tenant partition number to a loudspeaker paging zone	1072
Assigning a tenant partition number to the remote access extension	1073
Assigning a tenant partition number to a user extension	1073
Assigning a tenant partition number to a terminating extension group	1074

Assigning a tenant partition number to a trunk group	1075
Assigning a tenant partition number to a vector directory number	1075
Assigning the sources of music for the tenant partitions	1076
• Reports for Tenant Partitioning	1077
• Considerations for Tenant Partitioning	1077
• Interactions for Tenant Partitioning	1078
Temporary Bridged Appearance	1083
• Detailed description of Temporary Bridged Appearance	1083
• Hardware requirements for Temporary Bridged Appearance	1083
• Administering Temporary Bridged Appearance	1084
Screens for administering Temporary Bridged Appearance	1084
• Reports for Temporary Bridged Appearance	1084
• Considerations for Temporary Bridged Appearance	1084
• Interactions for Temporary Bridged Appearance	1085
Terminal Translation Initialization	1087
• Detailed description of Terminal Translation Initialization	1087
Attendant consoles	1087
Data modules	1088
Voice and data telephones	1088
ISDN-BRI telephones	1088
Analog queue warning ports and external alert ports	1089
Security	1089
• Hardware requirements for Terminal Translation Initialization	1089
• Administering Terminal Translation Initialization	1089
Screens for administering Terminal Translation Initialization	1089
• Reports for Terminal Translation Initialization	1090
• Considerations for Terminal Translation Initialization	1090
• Interactions for Terminal Translation Initialization	1090
Terminating Extension Group	1093
• Detailed description of Terminating Extension Group	1093
• Hardware requirements for Terminating Extension Group	1093

• Administering Terminating Extension Group	1093
Screens for administering Terminating Extension Group	1094
• Reports for Terminating Extension Group	1094
• Considerations for Terminating Extension Group	1094
• Interactions for Terminating Extension Group	1094

Transfer 1097

• Detailed description of Transfer	1097
Pull Transfer	1097
Abort Transfer	1097
Transfer Recall	1098
Transfer Upon Hangup	1098
Trunk-to-Trunk Transfer	1098
Outgoing Trunk to Outgoing Trunk Transfer	1098
Emergency Transfer	1099
• Hardware requirements for Transfer	1099
• Administering Transfer	1099
Screens for administering Transfer	1099
• Reports for Transfer	1100
• Considerations for Transfer	1100
Transfer - Outgoing Trunk to Outgoing Trunk	1101
Transfer - Trunk to Trunk	1101
• Interactions for Transfer	1101
Outgoing Trunk to Outgoing Trunk Transfer (OTTOTT)	1102
Trunk to Trunk Transfer	1103
Emergency Transfer	1103

Trunk Flash 1105

• Detailed description of Trunk Flash	1105
• Hardware requirements for Trunk Flash	1106
• Administering Trunk Flash	1106
Screens for administering Trunk Flash	1106
• End-user procedures for Trunk Flash	1106
• Reports for Trunk Flash	1106
• Considerations for Trunk Flash	1107
• Interactions for Trunk Flash	1107

Uniform Dial Plan 1109

- Detailed description of Uniform Dial Plan 1109
 - Uniform Dial Plan example 1111
- Hardware requirements for Uniform Dial Plan 1113
- Administering Uniform Dial Plan 1114
 - Prerequisites for administering Uniform Dial Plan 1114
 - Screens for administering Uniform Dial Plan 1115
 - Administering the Uniform Dial Plan table 1116
 - Administering the route pattern 1117
 - Service/Feature 1124
 - Administering the Node Number Routing table 1126
 - Administering the AAR Digit Conversion table 1126
 - Administering the ARS Digit Conversion table 1128
 - Administering the AAR Digit Analysis table 1129
 - Administering the ARS Digit Analysis table 1132
 - Administering the extension number portability numbering plan 1134
- Reports for Uniform Dial Plan 1135
- Considerations for Uniform Dial Plan 1135
- Interactions for Uniform Dial Plan 1136

Visually Impaired Attendant Service 1137

- Detailed description of Visually Impaired Attendant Service 1137
- Hardware requirements for Visually Impaired Attendant Service 1138
- Administering Visually Impaired Attendant Service 1138
 - Prerequisites for administering Visually Impaired Attendant Service 1138
 - Screens for administering Visually Impaired Attendant Service 1138
- Reports for Visually Impaired Attendant Service 1139
- Considerations for Visually Impaired Attendant Service 1139
- Interactions for Visually Impaired Attendant Service 1139

Voice Message Retrieval	1141
• Detailed description of Voice Message Retrieval	1141
• Hardware requirements for Voice Message Retrieval	1141
• Administering Voice Message Retrieval	1141
Screens for administering Voice Message Retrieval	1142
• Reports for Voice Message Retrieval	1142
• Considerations for Voice Message Retrieval	1142
• Interactions for Voice Message Retrieval	1143
 Whisper Paging	 1145
• Detailed description of Whisper Paging	1145
Call redirection overrides	1145
Group answering environments	1145
Network restrictions	1146
Speakerphones	1146
• Hardware requirements for Whisper Paging	1146
• Administering Whisper Paging	1146
Prerequisites for administering Whisper Paging	1146
Screens for administering Whisper Paging	1147
• End-user procedures for Whisper Paging	1147
Making whisper pages	1147
Allowing users to answer whisper pages quickly	1147
Allowing users to block whisper pages	1148
• Reports for Whisper Paging	1148
• Considerations for Whisper Paging	1148
• Interactions for Whisper Paging	1149
 Wideband Switching	 1151
• Detailed description of Wideband Switching	1151
Channel allocation	1152
Typical uses	1152
Endpoint applications with signaling	1153
ISDN-PRI terminal adapters	1153
Line-side (T1 or E1) ISDN-PRI facilities	1154
PRI endpoints	1154
Universal digital signal level 1 board	1154

Nonsignaling endpoint applications	1154
Data service unit/channel service unit	1154
Line-side (T1 or E1) facility	1155
Wideband access endpoint	1155
Guidelines and examples	1155
High-speed video conferencing	1155
Data backup connection	1156
Scheduled batch processing	1156
Primary data connectivity	1156
Networking	1156
ISDN-PRI trunk groups and channel allocation	1157
Facility lists	1157
Direction of trunk/hunting within facilities	1157
H11	1158
H12	1158
H0	1159
N x DS0	1159
Glare prevention	1160
Blocking prevention	1161
• Hardware requirements for Wideband Switching	1162
• Administering Wideband Switching	1162
Screens for administering Wideband Switching	1162
• Reports for Wideband Switching	1162
• Considerations for Wideband Switching	1163
• Interactions for Wideband Switching	1163
World Class Routing	1165
• Detailed description of World Class Routing	1165
Overview of automatic routing	1166
Understanding ARS analysis	1167
Examples of digit conversion	1168
ARS dialing without FAC	1169
Extensions	1170
AAR and ARS partitioning	1171
• Hardware requirements for World Class Routing	1171

• Administering World Class Routing	1171
Prerequisites for administering World Class Routing	1172
Screens for World Class Routing	1172
Managing calling privileges using COR and FRL	1173
Assigning the ARS FAC	1175
Setting up a location ARS FAC	1177
Prerequisites	1177
Displaying ARS analysis information	1178
Defining call types	1180
Defining operator-assisted calls	1180
Defining interexchange carrier calls	1181
Using restricted area codes and prefixes	1181
Using wildcards	1182
Defining local information calls	1183
Modifying call routing	1183
Adding a new area code or prefix	1183
Using ARS to restrict outgoing calls	1185
Defining ARS partitions	1186
Prerequisites	1186
Setting up partition groups	1187
Assigning a telephone to a partition group	1188
Setting up Time of Day Routing	1190
Prerequisites	1190
• Reports for World Class Routing	1192
• Considerations for World Class Routing	1192
• Interactions for World Class Routing	1193

Index	1195

Abbreviated Dialing

Use the Abbreviated Dialing (AD) feature to reduce the number of digits that you must dial to place a call. Instead of dialing the entire number, you dial a short code to access the number. The system then dials the stored number automatically. You can also assign abbreviated dialing buttons to telephones, so that you can dial frequently called numbers by pressing a single button.

Abbreviated Dialing is sometimes called speed dialing.

You can store telephone numbers in four different types of abbreviated dialing lists:

- Personal
- Group
- System
- Enhanced

You can also assign privileged numbers to abbreviated dialing lists. Users can use privileged numbers to place calls to numbers that might otherwise be restricted.

Privileged group-number, system-number, and enhanced-number lists provide access to numbers that usually might be restricted.

The switch type and version determine which lists are available and how many entries you can have on each list. You can assign up to 3 AD lists to each user or extension. You can assign any combination of a system list, an enhanced list, and as many as three personal lists or three group lists. The list can also have three group lists. Each entry on an abbreviated dialing list can have as many as 24 characters.

Detailed description of Abbreviated Dialing

This section provides a detailed description of the Abbreviated Dialing feature.

Abbreviated Dialing supports the following types of lists:

- Personal lists

Use personal lists for 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 telephone numbers to personal lists. A personal list is created automatically when you assign the list to an individual telephone. Each user can have as many as three personal lists. You cannot assign a personal list to the attendant.

- Group lists

Define group lists for groups or departments where members of the group must frequently dial the same numbers. You determine which users have access to group lists. Each user can have access to up to three group lists. You can program the list or you can designate a user in each group to program the list. 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 each telephone and allow everyone in the organization to use the list. If you allow everyone to use the system list, you should add only numbers to the list that anyone in your organization have permission to call. For example, you might want to add an emergency telephone number or telephone numbers for other office locations to this list.

The system list can contain up to 100 entries and can be changed by a system administrator. Also, an administrator can customize the button labels displayed on Avaya™ 2420 and Avaya™ 4620 telephone sets for these system-list entries.

- Enhanced lists

You can use enhanced lists for telephone users, data-terminal users, and attendants who need more list entries than those allowed in group-number and system-number lists. Two enhanced-number lists are allowed per system, in addition to the system-number list. The enhanced list can contain any number or dial-access code. You administer the enhanced lists and determine which users can access the list.

Hardware requirements for Abbreviated Dialing

The Abbreviated Dialing feature requires the following hardware:

- None

Administering Abbreviated Dialing

The following steps are part of the administration process for the Abbreviated Dialing feature:

- [Adding Abbreviated Dialing lists](#)
- [Assigning telephones for group lists](#)

This section describes:

- Any prerequisites for administering the Abbreviated Dialing feature
- The screens that you use to administer the Abbreviated Dialing feature
- Complete administration procedures for the Abbreviated Dialing feature

Prerequisites for Abbreviated Dialing

You must complete the following actions before you can administer the Abbreviated Dialing feature:

- On the **Optional Features** screen, you must ensure that the **Abbreviated Dialing Enhanced List** field is enabled before you can program an enhanced Abbreviated Dialing list. This screen shows the features that are enabled according to your license file.

If this field is not set to **y**, you cannot administer the Abbreviated Dialing feature. Contact your Avaya representative for assistance.

To view the *Optional Features* screen, type **display system-parameters customer-options**. Press **Enter**.

For a complete description of the many *Optional Features* screens, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Abbreviated Dialing

Screen name	Purpose	Fields
<i>Optional Features</i>	Ensure that the Abbreviated Dialing feature is enabled.	Abbreviated Dialing Enhanced List
<i>Abbreviated Dialing List</i>	Add Abbreviated Dialing lists.	• Size • Program Ext
<i>Station</i>	Define extensions for group lists.	Group

Adding Abbreviated Dialing lists

To add an Abbreviated Dialing group list:

- 1 Type **add abbreviated-dialing group next**. Press **Enter**.

The system displays the *Abbreviated Dialing List* screen ([Figure 1, Abbreviated Dialing List screen](#), on page 65). In this example, the next available group list is group 3.

Figure 1: Abbreviated Dialing List screen

add abbreviated-dialing group next		Page 1 of 1
Abbreviated Dialing List		
Group List: 3		Group Name: _____
Size (multiple of 5): ____		Program Ext: _____ Privileged? n
DIAL CODE		
01:	_____	
02:	_____	
03:	_____	
04:	_____	
05:	_____	

- 2 In the Group Name field, type a name for this list.
- 3 In the Size field, enter a number in multiples of 5. This number defines the number of entries on the dialing list.

For example, if you have eight telephone numbers that you want to store in the list, type **10** in the Size field.

- 4 In the **Program Ext** field, enter the extension.
- 5 Type the telephone numbers you want to store. Type one for each dial code.
Each telephone number can be up to 24 digits long.
- 6 Press **Enter** to save your changes.

You can display the new abbreviated dialing list to ensure that the information is correct, or print a copy of the list for your paper records.

Assigning telephones for group lists

To assign a telephone access to a group list:

- 1 Type **change station *n***, where *n* is the extension of the telephone that you want to assign to the group list. Press **Enter**.
The system displays the **Station** screen.
- 2 Press **NEXT** until you see the **Abbreviated Dialing** area ([Figure 2, Station screen](#), on page 66).

Figure 2: Station screen

change station 9876543		Page 3 of 3
Station		
SITE DATA		
Room: _____	Headset? <u>n</u>	
Jack: _____	Speaker? <u>n</u>	
Cable: _____	Mounting? <u>d</u>	
Floor: _____	Cord Length: <u>0</u>	
Building: _____	Set Color: _____	
ABBREVIATED DIALING		
List1: <u>group</u> <u>3</u>	List2: _____	List3: _____
BUTTON ASSIGNMENTS		
1: <u>call-appr</u>	4: _____	
2: <u>call-appr</u>	5: _____	
3: <u>call-appr</u>	6: _____	

- 3 Type **group** in any of the three **List** fields. Press **Enter**.
The system displays a blank list number field.
- 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 987-6543 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. You can also assign an abbreviated dialing button to this station. The user can then press one button to dial a specific number on one of their three assigned abbreviated lists of the user.

End-user procedures for Abbreviated Dialing

End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Abbreviated Dialing:

Using the Abbreviated Dialing program feature

These instructions apply to most Avaya telephones with display screens, and work with DEFINITY® software release 6.3 or later.

- 1** On your telephone, press the button labeled **Prog** to enter programming mode. If your telephone does not have the **Prog** softkey, press the **Menu** softkey and navigate to the **Prog** option.
The telephone goes off hook and enters the speaker phone mode.
- 2** Select the softkey/feature button you want to program until you see the label of the softkey you want to display. You will see the message **Change number? Yes = 1 No = 2**.
- 3** Pick the option you want. You will see a message **Enter number :** on the display. Enter the number you want that button to call.
- 4** Press the **#** key to save your changes.
- 5** You will see **Enter label** on the display. Use the dial pad to enter the label you want. The label can be up to five characters in length.
- 6** To exit from the **Prog** mode, hang up your telephone. Pressing the **Exit** button does not exit you from programming mode.

NOTE:

If the number you are entering for an Abbreviated Dialing button is an outside number, you must include 9 or any other applicable trunk code. Numbers programmed on softkeys can be up to 24 digits in length.

Reports for Abbreviated Dialing

The following reports provide information about the Abbreviated Dialing feature:

- None

Considerations for Abbreviated Dialing

This section provides information about how the Abbreviated Dialing feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Abbreviated Dialing feature under all conditions.

- You cannot remove a telephone or attendant if it is designated as the extension that is allowed 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 for Abbreviated Dialing

This section provides information about how the Abbreviated Dialing feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Abbreviated Dialing in any feature configuration.

- Last Number Dialed

The Last Number Dialed feature redials the same number a user just dialed. This happens even if the user accessed an abbreviated dialing list for the previous call. However, if the last dialed string includes any special characters, these characters are ignored by last-number-dialed call. Examples of special characters might be indefinite wait, mark, pause, suppress, wait and so on.

If the previously-called number was in an AD privileged list, and if the user is not usually 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.

Troubleshooting Abbreviated Dialing

Problem	Possible causes	Action
A user cannot access a abbreviated dial list.	The specific list might not be assigned to the user's telephone.	1 Type display station <i>n</i>, where <i>n</i> is the extension of the user. Press Enter. 2 Review the current settings of the List1, List2, and List3 fields to determine if the list is assigned to the telephone of the user.
	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. Press Enter 2 Verify that the user is dialing the appropriate feature access code.
	If the user attempted to press a feature button, they might have pressed the incorrect feature button.	1 Type display station <i>n</i>, where <i>n</i> is the extension of the user. Press Enter. 2 Review the current feature button assignments to determine if the user pressed 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 <i>n</i>, where <i>n</i> is the extension of the user. Press Enter. 2 Review the current feature button assignments to ensure if the list number and dial code are correct.
A user states that an abbreviated dialing list dials the wrong number.	- The user is using the wrong dial code. - The dial code is defined incorrectly. - ARS digit conversion was not administered correctly.	1 Ask the user the number that was dialed or the button that he or she pressed to determine which list and dial code the user 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 <i>n</i>, where <i>n</i> is a group list number. Press Enter.
		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 access the abbreviated dialing list again. Type the number. Press Enter.

Access Security Gateway

Use the Access Security Gateway (ASG) feature to prevent unauthorized access by requiring the use of the ASG Key for logging into the system. The ASG Key is a hand-held device that is used for accessing the system.

Detailed Description of Access Security Gateway

Use the Access Security Gateway (ASG) feature to authenticate authorized users as users attempt to access Communication Manager. Authentication is successful only when Communication Manager and ASG communicate with a compatible key. The challenge/response negotiation starts after a user establishes an RS-232 session, and enters a valid Communication Manager login ID. Communication Manager then issues a challenge based on the login ID the user enters, to which the user must provide the expected response. The core of this transaction is a secret key, which is information both the lock (ASG) and the key possess. The relevance of the authentication token used to perform the challenge and response negotiation is limited to the current session.

The supported key consists of a hand-held encryption-generating device called the ASG Key. The key device is preprogrammed with the appropriate secret key to communicate with corresponding ASG-protected login IDs on Communication Manager.

Hardware requirements for Access Security Gateway

The Access Security Gateway (ASG) feature requires the following hardware:

- A hand-held Access Security Gateway Key

Administering Access Security Gateway

The following steps are part of the administration process for the Access Security Gateway feature:

- [Setting up Access Security Gateway](#)
- [Disabling Access Security Gateway](#)
- [Restarting Access Security Gateway](#)
- [Modifying Access Security Gateway for a lost key](#)
- [Monitoring the Access Security Gateway History Log](#)

This section describes:

- Any prerequisites for administering the Access Security Gateway feature
- The screens that are required to administer the Access Security Gateway feature
- Complete administration procedures for the Access Security Gateway feature

Prerequisites for administering Access Security Gateway

You must complete the following actions before you can administer the Access Security Gateway feature:

- You must have a *superuser* login ID to administer the Access Security Gateway feature.
- View the **Optional Features** screen, and ensure that the Access Security Gateway field is set to **y**. If the Access Security Gateway field is set to **n**, your system is not enabled for the Access Security Gateway feature. Contact your Avaya representative before you continue with this procedure.

To access the **Optional Features** screen, type **change system-parameters customer-options**. Press **Enter**.

Screens for administering Access Security Gateway

Screen Name	Purpose	Fields
Optional Features	Ensure that the Access Security Gateway feature is on.	Access Security Gateway
Login Administration	Set up the ASG secret key.	• System Generated Secret Key or Secret Key • others optional fields as desired
Security-Related System Parameters	Set security-related parameter preferences.	All
Access Security Gateway Session History	View ASG session establishment and session rejection events.	All

Setting up Access Security Gateway

To set up Access Security Gateway:

- 1 Type **change login n**, where **n** is the alphanumeric *superuser* login ID. Press **Enter**.

The system displays the **Login Administration** screen ([Figure 3, Login Administration screen](#), on page 73).

Figure 3: Login Administration screen

```
change login 1234

                                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 you type y, the system automatically displays the *Access Security Gateway Login Administration* screen (page 2).
- 4 Perform one of the following actions:
 - In the System Generated Secret Key field, type y for a system-generated secret
 - In the Secret Key field, type your secret key.
- 5 Complete any of the remaining, optional fields on page 2 that you want to administer.
- 6 Press **Enter** to save your changes.
- 7 Type **change system-parameters security**. Press **Enter**.
The system displays the *Security-Related System Parameters* screen ([Figure 4, Security-Related System Parameters screen](#), on page 74).

Figure 4: Security-Related System Parameters screen

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? y

MAINT? n SYS-PORT? n

- 8** In the Access Security Gateway Parameters section, determine which of the listed port type fields to set to **y**. In this example, the SYSAM-RMT field is 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.

- 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 *n***, where ***n*** is the alphanumeric login ID.
The system displays the **Login Administration** screen ([Figure 3, Login Administration screen](#), on page 73).
- 2** Click **Next** until the **Access Security Gateway Login Administration** page appears.
- 3** On the **Access Security Gateway Login Administration** page, set the **Blocked** field to **y**.
When you set the **Blocked** field to **y** you do not remove the login from the system, but do temporarily disable the login.
- 4** Press **Enter** to save your changes.

NOTE:

A superuser can disable and restart access for another superuser.

Restarting Access Security Gateway

To restart a temporarily disabled Access Security Gateway access for login:

- 1 Type **change login *n***, where *n* is the alphanumeric login ID. Press **Enter**.
The system displays the **Login Administration** screen ([Figure 3, Login Administration screen](#), on page 73).
- 2 Click **Next** until the **Access Security Gateway Login Administration** page appears.
- 3 On the **Access Security Gateway Login Administration** page, set the **Blocked** field to **n**.
- 4 Press **Enter** to save your changes.

Modifying Access Security Gateway for a lost key

To modify ASG if the user loses an ASG Key:

- 1 Modify any logins associated with the lost Access Security Gateway Key. See the *Access Security Gateway (ASG) Key Release 1.0 User's Guide* for more information on how to change your PIN.
- 2 If the login is no longer valid, type **remove login *n***, where *n* is the alphanumeric login ID, to remove the invalid login from the system. Press **Enter**.
- 3 To keep the same login, change the Secret Key that is associated with the login to a new value.
- 4 Using the new secret key value, rekey the 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 rejection events except when the **Access to INADS Port** field on the **Login Administration** screen is set to **y**. You must be a superuser to use the **list asg-history** command.

To access the log:

- 1 Type **list asg-history**. Press **Enter**.
The system displays the **Access Security Gateway** screen ([Figure 5, Access Security Gateway Session History screen](#), on page 76).

Figure 5: Access Security Gateway Session History screen

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 displays the following fields:

- **Date** — Contains the date of the session establishment or rejection in the mm/dd format where mm = month and dd = day.
- **Time** — Contains the time of the session establishment or rejection in the hh/mm format where hh = hour and mm = minute.
- **Port** — Contains the mnemonic that is associated with the port on which the session was established or rejected. The port mnemonics for G3r systems are:
 - SYSAM-LCL
 - SYSAM-RMT
 - MAINT
 - SYS-PORT

For G3si systems, the port mnemonics are:

- MRG1
- INADS
- NET
- 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.

Reports for Access Security Gateway

- None

Considerations for Access Security Gateway

This section provides information about how the Access Security Gateway feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Access Security Gateway feature under all conditions.

- None

Interactions for Access Security Gateway

This section provides information about how the Access Security Gateway feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of the Access Security Gateway feature in any feature configuration.

- Customer Access to Initialization and Administration System (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 are denied access to the system through the SYSAM-RMT or INADS port, even if you use a valid ASG login ID.

If access to the INADS port is disabled for login, administering access to the SYSAM-RMT or INADS port via ASG does not override the INADS port restriction.

- Login Administration

ASG does not modify the standard user interface for Communication Manager login administration. Also, the standard Communication Manager login user interface is maintained in cases where ASG parameters are not administered for the login or the port.

- Security Violation Notification (SVN)

ASG does not support an interface to SVN. Session rejection events do not appear in the monitor security violations login report. 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 rejection events do not increment the Successful Logins, Invalid Attempts, Invalid IDs, Forced Disconnects, Login Security Violations, or Trivial Attempts counters that are maintained for the list measurements security-violations detail report. Additionally, ASG-related data is not included in login-specific information maintained by the measurements security violations summary report.

Administration Change Notification

Use the Administration Change Notification feature to notify adjunct systems when administration data on Avaya Communication Manager is changed. These notifications keep a client application that is running on an adjunct, such as Enterprise Directory Gateway, synchronized with Communication Manager.

Detailed Description of Administration Change Notification

Use the Administration Change Notification feature to allow Communication Manager to communicate with the Avaya Directory Enabled Management (DEM) client. The Administration Change Notification feature provides the client with real-time, integrated, directory-based, read and write access to Communication Manager administration data, based on rules defined by the customer. Administration Change Notification allows a client application to subscribe to notifications of changes to administration data in Communication Manager. This feature thus provides real-time updates whenever administration changes occur in a particular object, such as a station.

The Administration Change Notification feature tracks changes that are made through the System Access Terminal (SAT), INADS port, a Property Management System (PMS), a Call Management System (CMS), Avaya Site Administration, Avaya Network Administration, or Avaya Directory Gateway. The Administration Change Notification feature also tracks any changes that are made through a telephone interface, such as Terminal Translation Initialization (TTI), Personal Station Access (PSA), and Terminal Self Administration.

Communication Manager notifies the adjunct only that a data object is changed. To obtain details about the change, the adjunct must request this information from the server over a separate link.

Hardware requirements for Administration Change Notification

The Administration Change Notification feature requires the following hardware:

- None

Administering Administration Change Notification

The following steps are part of the administration process for the Administration Change Notification feature:

- [Initiating Administration Change Notification](#)

This section describes:

- Any prerequisites for administering the Administration Change Notification feature
- The screens that are required to administer the Administration Change Notification feature
- Complete administration procedures for the Administration Change Notification feature

Prerequisites for administering Administration Change Notification

You must complete the following actions before you can administer the Administration Change Notification feature:

- None

Screens for administering Administration Change Notification

Screen Name	Purpose	Fields
<i>Administration Changes</i>	View changes to administration data in Communication Manager.	All

Initiating Administration Change Notification

To initiate Administration Change Notification:

- 1 From the client application, type **notify history**. Press **Enter**.

The system displays the *Administration Changes* screen ([Figure 1, Administration Changes screen](#), on page 81).

Figure 1: Administration Changes screen

notify history					
ADMINISTRATION CHANGES					
Date	Time	Login	Actn	Object	Qualifier
3/16	14:21	Craft	Logn		

Communication Manager continues to send change notification over the Operations Support System Interface (OSSI) link until the command is canceled.

Reports for Administration Change Notification

The following reports provide information about the Administration Change Notification feature:

- None

Considerations for Administration Change Notification

This section provides information about how the Administration Change Notification feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Administration Change Notification feature under all conditions.

- None

Interactions for Administration Change Notification

This section provides information about how the Administration Change Notification feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of the Administration Change Notification feature in any feature configuration.

- None

Administered Connections

Use the Administered Connections (AC) feature to establish an end-to-end connection between two access or data endpoints. Communication Manager automatically establishes the connection based on the attributes that you administer. The Administered Connections feature provides the following capabilities:

- Support of both permanent and scheduled connections
- Auto restoration (preserving the active session) for connections that are routed over Software Defined Data Network (SDDN) trunks
- An administrable retry interval from 1 to 60 minutes per AC
- An administrable alarm strategy per AC
- An establish/retry/auto restoration order that is based on administered priority

Detailed description of Administered Connections

Use the Administered Connections (AC) feature to administer virtual private-line connectivity over the AT&T Switched Network. Access is over an ISDN trunk group for which the **Service Type** field on the **ISDN Trunk Group** screen is set to **SDDN** (Software Defined Data Network). 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.

You can establish an AC between the following endpoints:

- Two endpoints on the same server
- Two endpoints in the same private network, but on different servers
- One endpoint on the controlling server, and another endpoint off the private network

In all configurations, administer the AC on the server that has the originating endpoint. For an AC in a private network, if the two endpoints are on two different servers, Automatic Alternate Routing (AAR) through tie trunks (ISDN, DS1, or analog tie trunks) and intermediate servers are usually used to route the connection. If required, the connection can also be routed with 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 administered in the **Originator** field on the **Administered Connection** screen.

Access endpoints

Access endpoints are non signaling trunk ports. Access endpoints neither generate signaling to the far-end of the trunk nor respond to signaling from the far-end. You designate an access endpoint as the originating endpoint or the destination endpoint in an AC.

Typical applications

The following examples are typical AC applications:

- A local data endpoint that connects to a local or a remote access endpoint, such as:
 - a modular processor data model (MPDM) ACCUNET digital service that connects to SDDN over an ISDN trunk-group DS1 port; an MPDM
 - an MPDM ACCUNET digital service that connects to an ACCUNET Switched 56 Service over a DS1 port.
- A local-access endpoint that connects to a local or a remote access endpoint, such as a DSO cross-connect and a 4-wire leased-line modem to a 4-wire modem connection over 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.

Establishing Administered Connections

The originating server attempts to establish an AC only if one of the following conditions exist:

- AC is active.
- AC is due to be active (it is a permanent AC, or it is the administered time-of-day for a scheduled AC).
- The originating endpoint is in the 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 necessary state. The originating server uses the destination address to route the call to the desired endpoint. When the server establishes two or more ACs at the same time, the server 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 unavailable.
- Access is denied by Class of Restriction (COR), facilities restriction level (FRL), Bearer Capability Class (BCC), or an attempt is made to route voice-band data over SDDN trunks in the public switch network.
- The destination address is incorrect.
- The destination endpoint is busy.
- Other network or signaling failures occur.

In the event of a failure, an error is entered into the error log. This error generates an alarm, if your alarming strategy warrants an alarm. You can display AC failures with the **display status-administered connection** command. The originating server continues trying to establish an AC as long as an AC is scheduled to be active, unless the attempt fails because of an administrative error (for example, a wrong number) or service-blocking condition (for example, outgoing calls are barred).

- The administered retry interval of 1 to 60 minutes for each AC determines the frequency with which failed attempts are retried.
- Retries are made after the retry interval elapses, regardless of the restorable attribute of the AC.
- ACs are retried in priority order.
- When you change the time of day on the server, 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 scenarios 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 might be because of user action (in the case of a data endpoint), maintenance activity that results 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, such as a facility failure, occurs between the endpoints. If an AC drops because the AC was disabled, removed, or is no longer due to be active, no action is taken. If an AC drops because of changed AC attributes, the system makes an immediate attempt to establish the connection with the changed attributes, if the AC is still scheduled to be active. Existing entries in the error/alarm log are resolved if the entries no longer apply. If an AC involves at least one data endpoint, and handshake failure causes the connection to be dropped, no action is taken for that AC until you run the **change administered-connection** command.

Administered Connections failure: auto restoration and fast retry

When an active AC drops prematurely, you must invoke either auto restoration or fast retry for auto restoration to be attempted for an active AC. If you administer an AC for auto restoration and the connection was routed over SDDN trunks, auto restoration is attempted. During restoration, connections are maintained between the server and both endpoints. In addition to maintaining the active session, AC also provides a high level of security by prohibiting other connections from intervening in active sessions. Auto restoration usually completes before the 60-second endpoint holdover interval. If auto restoration is successful, the call might be maintained, but this is not guaranteed. 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 indicated by the **restored** value in the **Connection State** field on the **Administered-Connection Status** screen. Although a restoration is successful, the data session might not be preserved.

If auto restoration is not active, or if the AC is not routed over SDDN trunks, the server immediately attempts a fast retry to reestablish the connection. The server also attempts a retry if the originating endpoint caused the drop. With fast retry, connections are not maintained on both ends. Fast retry is not attempted for an AC that was last established with fast retry, unless that AC is active for at least 2 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 scheduled to be active.

Hardware requirements for Administered Connections

The Administered Connections feature requires the following hardware:

- None

Administering Administered Connections

The following steps are part of the administration process for the Administered Connections feature:

- [Setting up Administered Connections](#)

This section describes:

- Any prerequisites for administering the Administered Connections feature
- The screens that are required to administer the Administered Connections feature
- Complete administration procedures for the Administered Connections feature

Prerequisites for administering Administered Connections

You must complete the following actions before you can administer the Administered Connections feature:

- None

Screens for administering Administered Connections

Screen Name	Purpose	Fields
<i>Data Modules</i>		All
<i>DS1 Circuit Pack</i>		All
<i>Access Endpoint</i>		All
<i>Trunk Group</i>		All
<i>Class of Restriction</i>		All
<i>Class of Service</i>		All
<i>Dial Plan Record</i>		Local Node Number
<i>Administered Connection</i>		All
<i>Station</i>	assign one button as ac-alarm	Feature Button Assignments area
<i>Attendant Console</i>	assign one button as ac-alarm	Feature Button Assignments area

Setting up Administered Connections

NOTE:

For detailed information about the screens that you use in the following procedure, see the appropriate Feature Description, or the Screen Reference chapter of the *Administrator's Guide for Avaya Communication Manager*.

To set up an Administered Connection:

- 1 In the **Types** field on the **Data Modules** screen, choose one of the following data module Types. Administer all fields that the screen displays for that data module type.
 - Data Line Data Module. Use with Data Line circuit pack.
 - Processor/Trunk Data Module. Use with:
 - MPDMs, 700D, 7400B, 7400D, or 8400B
 - MTDMs, 700B, 700C, 700E, or 7400A
 - Processor Interface Data Module. See *Administration for Network Connectivity for Avaya Communication Manager* for more information.
 - X.25 Data Module. See *Administration for Network Connectivity for Avaya Communication Manager* for more information.
 - 7500 Data Module. Use with an 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. Use with server node carriers.
- 3 On the **Access Endpoint** screen, administer all fields.
- 4 In the **Group Type** field on the **Trunk Group** screen, choose one of the following Group Types. Administer all fields that the screen displays for that group type.
 - ISDN-BRI
 - ISDN-PRI
 - Tie
- 5 On the **Class of Restriction** screen, administer all fields.
- 6 On the **Class of Service** screen, administer all fields.
- 7 On the **Dial Plan Record** screen, administer the **Local Node Number** field with a number that matches the DCS (distributed communications system) server node number and the CDR (Call Detail Recording) node number. Valid values are 1 to 63.
- 8 On the **Administered Connection** screen, administer all fields.
- 9 On the **Station** screen, assign one button as **ac-alarm**.
- 10 On the **Attendant Console** screen, assign one button as **ac-alarm**.

Reports for Administered Connections

The following reports provide information about the Administered Connections feature:

- None

Considerations for Administered Connections

This section provides information about how the Administered Connections feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Administered Connections feature under all conditions.

- None

Interactions for Administered Connections

This section provides information about how the Administered Connections feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of the Administered Connections feature in any feature configuration.

- Automatic Alternate Routing (AAR)/Automatic Route Selection (ARS)/Generalized Route Selection (GRS)
Use these features when routing an AC.
- Abbreviated Dialing
Use Abbreviated Dialing entries in the **Destination** field on the **Administered Connections** screen. Entries must comply with restrictions.
- Busy Verification of stations and trunks
This feature does not apply to access endpoints because access endpoints are used only for data.
- Call Detail Recording (CDR)
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 (COR)
Reserve a COR for AC endpoints and SDDN trunks to restrict endpoints that are not involved in AC from connecting to SDDN trunks or endpoints that are involved in AC.
- Class of Service (COS)/Call Forwarding
Assign a COS that blocks Call Forwarding activation at the AC endpoint.
- Data Call Setup
Do not assign a default dialing destination to a data module when the data module is used in an AC.
- Data Hotline
Do not assign a hotline destination to a data module when the data module 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 DMI endpoints 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 originating endpoint of an AC.
- **Modem Pooling**
If you require a modem in an AC, a modem 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 might fail if the only backup route is over the facility on which the backup D-channel is administered. The backup D-channel might not come into service in time to handle the restoration attempt.
- **Set Time Command**
When you use the **set time** command change the system time, 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 server attempts to establish the AC.
If any AC, scheduled or continuous, is in retry mode and the system time changes, the server 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 that are used in an AC to prevent display of call-processing messages, such as 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.

Administrable Language Displays

Use the Administrable Language Displays feature to display telephone messages in one of six languages. The meanings of the messages do not change, only the language.

Detailed description of Administrable Language Displays

With the Administrable Language Displays feature, you can select a language for the static messages that telephones and attendant consoles display. The system includes approximately 900 static display messages, such as “Transfer complete.” You can choose one of six languages: English, French, Italian, Spanish, user-defined, or Unicode.

The following types of information are pre-translated for display in English, French, Italian, and Spanish. If you select **user-defined**, you must enter the translation for each message.

- 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

Hardware requirements for Administrable Language Displays

The Administrable Language Displays feature requires the following hardware:

- Telephones and attendant consoles that have 40-character displays. Each language character set requires specific telephones. Call your Avaya representative for details.
- To use Unicode Message files, you must have Unicode-capable stations. In the November 2003 Release of Communication Manager, the 4620SW and Avaya Softphone R5.0 are supported. Only these Unicode-capable stations have the script (font) support that is required to match the scripts that the Unicode Message file uses.

Administering Language Displays

The following steps are part of the administration process for the Administrable Language Displays feature:

- [Setting the display language](#)
- [Entering translations for a user-defined language](#)
- [Administering Unicode display](#)

This section describes:

- Any prerequisites for administering the Administrable Language Displays feature
- The screens that are required to administer the Administrable Language Displays feature
- Complete administration procedures for the Administrable Language Displays feature

Prerequisites for administering Administrable Language Displays

You must complete the following actions before you can administer the Administrable Language Displays feature:

- Ensure that the **Display Character Set** field on the **System Parameters Country-Options** screen is set to the character type that you want to display. This field is set by Avaya. If the **Display Character Set** field is not set to the desired character type, contact your Avaya representative.
- Ensure that the type of telephone your company uses has a 40-character display, and supports the characters that you want to display. Each character set requires specific telephones. Contact your Avaya representative for details.

- For Unicode display, ensure that the Communication Manager (CM) phone messages for the Unicode Display language are loaded on CM. The file *avaya_unicode.txt* contains Avaya Communication Manager phone messages. The file *custom_unicode.txt* contains Avaya Communication Manager posted messages and system labels. These files can be loaded at upgrade/installation, or can be downloaded at any time from the Avaya support Web site (<http://www.avaya.com/support>).

The Avaya Excel Translation Editor for creating Unicode translations requires Microsoft Excel 2000 or above.

- In order to support Unicode on the 4620SW IP terminal, you must:
 - install the appropriate version of firmware on the set
 - set the language as desired

For more information on administering terminals, see the *4600 Series IP Telephone R2.0 LAN Administrator's Guide* (555-233-507), and the *4620/4620SW IP Telephone R2.0 User's Guide* (555-233-781).

Screens for administering Administrable Language Displays

Screen Name	Purpose	Fields
<i>System Parameters - Country Options</i>	Ensure that the appropriate character set is enabled for the language that you want to use.	Display Character Set
<i>Attendant Console</i>	Set the display language.	Display Language
<i>Language Translations</i>	Enter user-defined translations of messages.	Translation

Setting the display language

To set the display language:

- 1 Type **change attendant *n***, where *n* is the number of the attendant console that you want to change. Press **Enter**.

The system displays the *Attendant Console* screen ([Figure 2, Attendant Console screen](#), on page 94).

Figure 2: Attendant Console screen

change attendant 1
ATTENDANT CONSOLE 1
Page 1 of 3

Type: console
 Extension: 1000
 Console Type: principal
 Port: 01C1106

Name: Gunther
 Group: 1
 TN: 1
 COR: 1
 COS: 1

Auto Answer: none
 Data Module? n
 Disp Client Redir? n
 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 Display Language field, enter the display language you want to use.

NOTE:

The **Display Language** value **unicode** is available only for station type 4620SW or 4610SW.

- 3 Press **Enter** to save your changes.

Entering translations for a user-defined language

To enter translations for a user-defined language:

- 1 Type **change attendant *n***, where *n* is the number of the attendant console you want to change. Press **Enter**.
The system displays the **Attendant Console** screen ([Figure 2, Attendant Console screen](#), on page 94).
- 2 In the Display Language field, enter **user-defined**.
- 3 Press **Enter** to save your changes.
- 4 Type **change display-messages *n***, where *n* is the message for which you want to translate the display language. Click **help** to view the messages that you can choose to translate. Press **Enter**.
The system displays the **Language Translations** screen for the type of message that you want to translate. ([Figure 3, Language Translations screen](#), on page 95). This example shows the screen for message type “Transfer-conference.”

Figure 3: Language Translations screen

change display-messages transfer-conference	Page 1 of 3
Language Translations	
1. English: Transfer completed. Translation: abtretung abgeschlossen	

- 5 In the **Translation** field, type the translation of the message in the user-defined language. In this example, the translation is **abtretung abgeschlossen**.
- 6 Press **Enter** to save your changes.

For more information on telephone displays, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Administering Unicode display

To use Unicode display languages, you must have the appropriate Avaya Unicode Message files loaded on Communication Manager. These files are named *avaya_unicode.txt* and *custom_unicode.txt*. In the November 2003 2.0 Release of Communication Manager, there are 5 languages available: English, Japanese, Chinese, Russian, and German. The Unicode Message files for each of these languages are available for download on the Avaya support Web site (<http://www.avaya.com/support>).

NOTE:

Message files for each of these languages are contained in a downloadable ZIP file that also contains the following two white papers: *CM2.0WhitePaper.pdf* and *UnicodeMessageFileWhitePaper.pdf*. Refer to these white papers for details on the following procedures:

To administer Unicode display:

- 1 Download the appropriate Unicode message file to your PC. For an existing translation, download the desired language. To create a new translation, download the English version.
- 2 If necessary, create a new translation, or modify an existing translation. Use the Avaya Excel Translation Editor to create or modify translations.
- 3 Use the AvayaL10n Tool to prepare the Unicode Message file for installation. This tool is included in the Unicode Message file ZIP file. The Unicode Message file will *not* install if you do not first process the file with this tool.
- 4 Transfer the Unicode Message file to an Avaya media server that is running Communication Manager 2.0 or later. You can use the Avaya Web pages, the Avaya Installation Wizard, or ftp to transfer the Unicode Message file.
- 5 Install the Unicode message file. You can install Unicode Message files from either the Install Web page or the Avaya Installation Wizard. Note that the Installation Wizard is the same wizard that you use to transfer Unicode Message files to an Avaya media server that is running Communication Manager.

- 6 Load the Unicode message files. After you install the Unicode Message files, you must do a reset 4 to load the Unicode Message files in the Avaya media server that is running Communication Manager 2.0 or later. Starting with Communication Manager 2.1, a reset 4 is no longer required to load the Unicode Message file *custom_unicode.txt*.
- 7 Type **change attendant *n***, where *n* is the number of the attendant console you want to change. Press **Enter**.

The system displays the **Attendant Console** screen ([Figure 2, Attendant Console screen](#), on page 94).
- 8 In the Display Language field, enter **unicode**.
- 9 Press **Enter** to save your changes.

Reports for Administrable Language Displays

- None

Considerations for Administrable Language Displays

This section provides information about how the Administrable Language Displays feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Administrable Language Displays feature under all conditions.

- None

Interactions for Administrable Language Displays

This section provides information about how the Administrable Language Displays feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of the Administrable Language Displays feature in any feature configuration.

- None

Troubleshooting Administrable Language Displays

This section lists the known or common problems that users might experience with the Administrable Language Displays feature.

Problem	Possible cause	Action
The characters that appear on the display are not what you entered.	This feature is case sensitive.	Check the table to ensure that you entered the characters in the correct case.
You entered ~c, but an asterisk (*) appears on the display instead.	Lowercase c has a specific meaning in Avaya Communication Manager. Therefore you cannot map c to any other character. An asterisk (*) appears in its place.	Use a different letter for this character mapping.
You entered ~-> or ~<- but 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 these characters.	Check the model of the keyboard that you are using.
Enhanced display characters appear in fields that you did not update.	If an existing display field contains a tilde (~) that is followed by Roman characters, and you update and submit that screen, that field displays the enhanced character set.	Remove the existing tilde before you submit the screen.
The terminal displays nothing at all.	Some unsupported terminals do not display anything if a special character is presented.	Check the model of the display terminal that you are using.
You entered a character with a descender and part of the descender 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.	Use Group1 equivalents for the letters g, j, p, q, and y.

Administration Without Hardware

Use the Administration Without Hardware (AWOH) feature to administer telephones before you actually install the telephones or move the telephones. When you move the telephones, the system preserves features that the users activate, for example, the Call Forwarding feature.

Detailed description of Administration Without Hardware

This section provides a detailed description of the Administration Without Hardware (AWOH) feature.

With AWOH you can enter telephone translations without assigning ports. Therefore, AWOH streamlines system initialization, major additions, and rearrangements or changes. You can add or change a **Station** screen, and you can store duplicated telephones, without specifying a port location.

Physical characteristics of an AWOH telephone

AWOH stations cannot generate alarms or errors, because the physical telephones or data terminals are not associated with a **Station** screen. Neither the system nor other telephones or data terminals can affect the lamp or the alerting tones of an AWOH telephone. If a user presses a button on telephone or a data terminal that does not have an associated **Station** screen, the action has no effect on system operation.

User activated features

When someone moves a telephone or a data terminal the user-activated features, such as Call Forwarding and Send All Calls, remain active. Any action that changes the lamps or the status of the station is reflected when the telephone or data terminal is again associated with a **Station** screen.

Association and disassociation

Telephone users, data-terminal users, and technicians can disassociate telephone translations from the current terminal port, and then associate the telephone translations with another terminal port. Users use the Personal Station Access (PSA) feature to disassociate and associate telephones. Technicians use the Terminal Translation Initialization (TTI) feature to disassociate and associate telephones.

An AWOH telephone is considered to be disassociated when no hardware port is assigned to the telephone. To indicate that no hardware is associated with a telephone, type **x** in the **Port** field of the appropriate screen. When the port is assigned later, the AWOH telephone is considered to be associated.

Phantom extensions

You can use phantom extensions to provide call coverage, including AUDIX coverage, for users who do not have telephones or data terminals that are physically located on the system.

You can also use phantom extensions for the automatic call distribution (ACD) Dialed Number Identification Service (DNIS). With ACD DNIS you can administer a phantom extension on the switch for each type of call that ACD agents need to identify.

To use ACD DNIS, either a user with console permissions forwards the phantom extension to an ACD split, or the coverage path of the phantom extension includes an ACD split. The Name field for the phantom extension identifies the service that the caller wants. The agent then uses this information to address the caller properly.

Hardware requirements for Administration Without Hardware

The Administration Without Hardware feature requires the following hardware:

- None

Administering Administration Without Hardware

The following steps are part of the administration process for the Administration Without Hardware (AWOH) feature:

- [Assigning AWOH for a hunt-group queue](#)
- [Assigning AWOH to a telephone](#)
- [Assigning AWOH to an attendant console](#)
- [Assigning AWOH to a data module](#)

This section describes:

- Any prerequisites for administering the Administration Without Hardware feature
- The screens that you use to administer the Administration Without Hardware feature
- Complete administration procedures for the Administration Without Hardware feature

Prerequisites for administering Administration Without Hardware

You must complete the following actions before you can administer the Administration Without Hardware feature:

- None

Screens for administering Administration Without Hardware

Screen name	Purpose	Fields
<i>Attendant Console</i>	Assign a port to the attendant console.	Port
<i>Data Module</i>	Assign a port to the data module.	Port
<i>Hunt Group</i>	Assign port information for a queue.	• Calls Warning Port • Time Warning Port
<i>Station</i>	Assign a port to a station.	Port

Assigning AWOH for a hunt-group queue

To assign AWOH for a hunt-group queue:

- 1 Type **change hunt-group *n***, where *n* is the number of the hunt group to which you want to assign AWOH. Press **Enter**.

The system displays the *Hunt Group* screen ([Figure 4, Hunt Group screen](#), on page 101).

Figure 4: Hunt Group screen

change hunt-group 1		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: <u>x</u> ____	Extension: ____	
Time Warning Threshold: ____	Port: <u>x</u> ____	Extension: ____	

- 2 In the Calls Warning Port field, type **x**.

Note that the system displays the Calls Warning Port field if the Queue field is set to **y**.

- 3 When you type **x** in the In the Calls Warning Port field, the system displays an Extension field.

In the Extension field, type an extension that a technician can use, along with TTI, to assign a port number from the actual port. When a technician or an administrator assigns a port number, the system removes the extension number and the extension number becomes unassigned. An administrator uses the **change hunt-group** command to assign the port number.
- 4 In the Time Warning Port field, type **x**.

Note that the system displays the Time Warning Port field if the Queue field is set to **y**.
- 5 When you type **x** in the In the Time Warning Port field, the system displays an Extension field.

In the Extension field, type an extension that a technician can use, with TTI, to assign a port number from the actual port. When a technician or an administrator assigns a port number, the system removes the extension number and the extension number, becomes unassigned. An administrator uses the **change hunt-group** command to assign the port number.
- 6 Press **Enter** to save your changes.

Assigning AWOH to a telephone

To assign AWOH to a telephone:

- 1 Type **change station *n***, where ***n*** is the number of the extension to which you want to assign AWOH. Press **Enter**.

The system displays the **Station** screen ([Figure 5, Station screen](#), on page 102).

Figure 5: Station screen

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?	
Model:		
	Media Complex Ext:	
	IP Softphone? y	
	Remote Office Phone? y	

- 2 In the Port field, type **x**.
- 3 Press **Enter** to save your changes.

Assigning AWOH to an attendant console

To assign AWOH to an attendant console:

- 1 Type **change attendant *n***, where *n* is the number of the attendant console to which you want to assign AWOH. Press **Enter**.

The system displays the *Attendant Console* screen ([Figure 6, Attendant Console screen](#), on page 103).

Figure 6: Attendant Console screen

change attendant 1Page 1 of 3

ATTENDANT CONSOLE 1

Type: console

Extension: 1000

Console Type: principal

Port: 01C1106

Security Code:

Name: 27 character attd cons name

Group: 1

TN: 1

COR: 1

COS: 1

Auto Answer: none

Data Module? y

Disp Client Redir? n

Display Language: english

H.320 Conversion? n

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)

Local Remote

Local Remote

Local Remote

1: 9

2: 82

3:

4:

5:

6:

7:

8:

9:

10:

11:

12:

HUNDREDS SELECT BUTTON ASSIGNMENTS

1:

2:

3:

4:

5:

6:

7:

8:

9:

10:

11:

12:

13:

14:

15:

16:

17:

18:

19:

20:

- 2 In the Port field, type **x**.
- 3 Press **Enter** to save your changes.

Assigning AWOH to a data module

To assign AWOH to a data module:

- 1 Type **change data-module *n***, where *n* is the number of the data module to which you want to assign AWOH. Press **Enter**.

The system displays the *Data Module* screen ([Figure 7, Data Module screen](#), on page 104).

Figure 7: Data Module screen

add data-module next		Page 1 of 1	
DATA MODULE			
Data Extension: 502	Name:	BCC: 2	
Type: pdm	COS: 1	Remote Loop-Around Test? n	
Port:	COR: 1	Secondary data module? n	
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 (first 26 characters)		
1:			

- 2 In the Port field, type x.
- 3 Press **Enter** to save your changes.

Reports for Administration Without Hardware

The following reports provide information about the Administration Without Hardware feature:

- None

Considerations for Administration Without Hardware

This section provides information about how the Administration Without Hardware feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Administration Without Hardware under all conditions. The following considerations apply to Administration Without Hardware:

- None

Interactions for Administration Without Hardware

This section provides information about how the Administration Without Hardware (AWOH) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of AWOH in any feature configuration.

- Association and disassociation interactions
 - Attendant

An attendant cannot disassociate the attendant station if the attendant has a call in a queue, a call on hold, or a call that is active. An attendant cannot dissociate the attendant station under those conditions, even if the attendant is in Position Available mode.
 - Attendant Night Service

A night service station cannot be removed while the station is in night service. See the “Night Service” feature for more information.
 - Attendant Release Loop Operation

The system reclassifies calls as attendant group calls if the attendant holds the calls with the release loop operation, and if the attendant disassociates the attendant station before the attendant timed-reminder interval expires.
 - Automatic Callback

If a station becomes disassociated while another station has automatic callback active for the disassociated station, the automatic callback light turns off. The callback sequence does not occur for the station that has automatic call back active.
 - Bridged Call Appearance

If a station has a bridged call appearance of an off-hook station, the first station can disassociate at any time, and not disrupt a call in progress on the bridge.

If a station with a bridged call appearance associates itself while the extension for the bridged appearance is on a call, the associated station can join the call.

You cannot disassociate a station from a bridged call appearance. You must disassociate a station from the port on which the station resides.
 - Call Coverage

If an AWOH station disassociates while Send All Calls or Go to Coverage is active, both features remain active.
 - Call Coverage Answer Group

If a technician or an administrator associates a station, the station cannot join calls that are in progress at the call-coverage answer group. The station can join subsequent calls.
 - Call Forward

A station can disassociate while Call Forwarding is active.

If a Call Forwarding destination disassociates, Call Forwarding to that extension remains active.
 - Call Park

If a line appearance is available, a user can disassociate a while a call to that station is parked. The user can then retrieve the call from another station.

— Call Pickup

If a line appearance is available, a member of a Call Pickup group can disassociate a station at any time.

If a call is in progress to any extension in a pickup group, any member of the group can disassociate or associate a station. The pickup group member who dissociates or associates a station, does not join the group for the call in progress. The member can participate in subsequent calls.

— Customer-premises equipment (CPE) Alarm

If a station that is administered with a CPE alarm becomes associated with a port while an alarm is active, the station receives the alarm when the station is associated.

— Hunt Group with Uniform Call Distribution (UCD) and Direct Department Calling (DDC)

If a technician or an administrator associates a station, the station cannot join calls that are in progress at the hunt group. The station can join subsequent calls.

— Hold

A station user can place a call on hold. The user can then disassociate the station, associate the station, and retrieve the call that the user placed on hold.

— Intercom Group - Auto/Dial

See the “Data Call Setup” feature for more information.

— Message Light

You do not need to delete messages before you dissociate a station. If a station receives messages while it is disassociated, the system updates the message light when someone associates the station.

— Send All Calls

Send All Calls remains active when a station is disassociated.

— Station-to-Station Call

You cannot disassociate a station while someone is active on a call at the station.

— Terminating Extension Group

If a technician or an administrator associates a station, the station cannot join calls that are in progress at the terminating extension group (TEG). The station can join subsequent calls.

— Transfer

After a user A successfully uses the Transfer feature to connect a caller with user B, user A can dissociate the station. The system processes the call between the caller and the transferred-to user as a station-to-station call.

— Trunk Group Night Service

You disassociate a night service destination that is a station the same way that you disassociate a station that is not a night service destination. See the “Data Call Setup” feature for more information.

You disassociate a night service destination that is an attendant the same way that you disassociate an attendant that is not a night service destination. See the “Attendant” features for more information.

- Attendant interactions

- Attendant Group

If all attendants of a group use AWOH consoles, internal callers receive ringback tone indefinitely. Attendant AWOH consoles operate the same way as AWOH stations in group interactions.

- Attendant Override

When an attendant activates Attendant Override, the attendant hears a busy signal for calls to AWOH extensions, and the attendant bypasses Call Coverage for the extensions.

- Attendant Return Call

If the attendant extends a call to a station, and then the attendant dissociates the attendant station before the system returns the call to the attendant, the system reclassifies the call. The system reclassifies the call as an attendant group call, and routes the call to an attendant group.

- Emergency Access to Attendant

A caller receives a busy signal if:

- The Emergency Access feature to the attendant is active.
 - All attendants use AWOH consoles.
 - No backup extension is administered.
 - The backup extension is also an AWOH extension.

- Interposition Calling, Attendant to Attendant

A caller receives the intercept tone for a call to an attendant who has an AWOH console.

- Night Station Service

A caller receives a busy signal if the attendant activates Night Station Service, and the night service endpoint in an AWOH extension.

- Serial Calling

With a serial call, the attendant is not in a busy state after the attendant releases a call. Because the attendant is not in a busy state, the attendant can disassociate the attendant station.

If the attendant extends a call to a station, and then the attendant dissociates before the system returns the call to the attendant, the system reclassifies the call. The system reclassifies the call as an attendant group call, and routes the call to an attendant group.

When an attendant attempts to extend a call to an AWOH extension, the attendant hears a busy signal.

- Data Modules

Users can use the following methods to associate and disassociate data modules:

- Data-terminal dialing
- Telephone dialing
- Other devices

The devices include the use of a default set type to associate, and then remove the default set type and replace the default set type with the proper data endpoint

Because digital-terminal data modules (DTDMs) reside on some station types, the port is automatically inherited from the host station. The DTDM receives a port identification when the station associates or disassociates.

- Administered Connections

If you administer a connection without hardware translation, the system attempts to establish a connection only when both endpoints are associated with hardware translation.

You disassociate an administered connections when you type **x** in the **Port** field on the **Data Module** screen. A technician uses Terminal Translation Initialization (TTI) to disassociate an administered connection.

- Hunt Group - Uniform Call Distribution and Direct Departmental Calling (UCD/DDC)

See the “Call Coverage” feature for more information.

- Terminal-to-Data Module Call

See the “Data Call Setup” feature for more information.

- Transfer

See the “Transfer” feature for more information.

- Data-terminal interactions

- Administered Connections

An administered connection endpoint can be an AWOH extension.

- Data Call Setup

- Data Terminal Dialing

A keyboard-dialed call that terminates to a data-endpoint that is an AWOH station, causes a BUSY message on the screen. A busy signal indicates that the terminal is in use, out of service, or AWOH.

- Telephone Dialing

See the “Data Call Setup” feature for more information.

- Hunt-Group (UCD/DDC)

See the “Call Coverage” feature for more information.

- Incoming Destination

If the incoming destination is an AWOH extension, the caller hears ringback tone from the central office (CO). The system routes incoming calls based on features that are active for the extension. These features include, for example, Call Forwarding and Call Coverage.

- Terminal-to-Data Module Call

If a data endpoint is an AWOH station, the caller either sees a BUSY message on the screen or hears a busy signal, depending on the originating hardware. See the “Data Call Setup” feature for more information.

- Telephone interactions
 - Abbreviated Dialing

AWOH does not change any aspect of the Abbreviated Dialing feature.

A station that has Abbreviated Dialing active, and then becomes disassociated, retains the abbreviated dialing list entries.
 - Automatic Call Distribution

A user cannot log an AWOH station into an Automatic Call Distribution (ACD) split. The user can log on directly or use a Bridged Call Appearance.
 - Automatic Callback

A user cannot activate the Automatic Callback feature for a call to an AWOH extension. If a user attempts to use the Automatic Callback feature for a call to an AWOH extension, the system sends the reorder tone to the user.
 - Bridged Call Appearance

A user can use a bridged call appearance of an AWOH extension to place a call.

A user can use a bridged call appearance of an AWOH extension to answer a call to the AWOH telephone.

An AWOH extension can contain a bridged call appearance, but a user cannot use the bridged call appearance on the AWOH telephone to place a call.
 - Busy Verification of Terminals and Trunks

When you use busy verification of terminals and trunks on an AWOH extension, the telephone appears to you as an out-of-service telephone.
 - Call Coverage

AWOH telephones interact with Call Coverage as if all call appearances are busy.

Call Coverage can be active at a disassociated station.
 - Call Forward

Call Forwarding can be active at an AWOH station, while the station is in a disassociated state. When the extension is associated, Call Forwarding is active.
 - Call Park

A call to an AWOH extension can be parked only if the primary extension has a bridged call appearance on a non-AWOH telephone. A call that is parked from a bridged call appearance is parked on the primary extension.
 - Call Waiting Termination

Call Waiting Termination can be administered on a single-line AWOH extension, but the caller receives a busy signal if the extension is disassociated.
 - Conference

Users cannot include an AWOH extension in a conference call.
 - Customer-Provided Equipment (CPE) Alarm

If a CPE alarm is active for an AWOH extension, no equipment is available to ring or light.
 - Data Buttons

Data buttons are not lit for AWOH data modules.

— Display

The AWOH feature does not change the display for calls that originate or terminate from a bridged call appearance for an AWOH extension.

— Facility Busy Indication

A telephone can have a busy indicator light for an AWOH extension, but the light is not lit. The light is not lit because Facility Busy Indication indicates whether an extension is off-hook or on-hook, even if you assign bridged appearances. An AWOH extension is always on-hook.

You can administer a busy indicator light on AWOH extensions. When an administrator or a technician assigns a port to the extension, the busy indicator light functions as usual.

— Incoming Destination

If the incoming destination is an AWOH extension, the caller hears ringback tone from the CO. The system routes incoming calls based on features that are active for the extension, such as Call Forward and Call Coverage.

— ISDN-BRI voice terminals

If TTI is enabled, you cannot use the SAT to enter **x** in the **Port** field of a BRI telephone **Station** record that is already connected to the switch. Instead, you must dial the TTI disassociate code from the telephone. For more information, see the “Terminal Translations Initialization (TTI)” feature.

— Leave Word Calling

You can leave a leave-word-calling message at an AWOH extension.

— Manual Message Waiting

When a user activates Manual Message Waiting toward an AWOH extension, there is no telephone on which to light the message waiting lamp. However, once the AWOH extension is associated with a port or telephone, the message waiting lamp lights.

— Manual Signaling

Manual Signaling to an AWOH extension has no effect on system operation, because no terminal exists to signal.

The system does not send a message to the originator of the call to inform the caller that the extension is an AWOH extension, and therefore cannot receive the signal.

— Personal Central Office Line (PCOL)

You can administer AWOH extensions with PCOL. If a call terminates at an AWOH extension that does not have Call Coverage active, the caller receives the ringback tone. The ringback tone indicates that the call is unanswered. If the AWOH extensions has Call Coverage active, the system routes the call to coverage.

— Priority Calling

A user who sends a priority call to an AWOH extension hears a busy signal.

— Send All Calls

Send All Calls remains active on AWOH extensions.

— Station Hunting

You can assign an AWOH extension to a station hunting chain.

— Station-to-Station Call

The system processes a call to an AWOH extension in the same way that the system processes a call to a telephone for which all call appearances are busy.

— Transfer

The system processes a transfer to an AWOH station in the same way that the system processes a call to a station that is busy.

Alphanumeric Dialing

Use the Alphanumeric Dialing feature to allow users to enter a alphanumeric name to place data calls instead of a numeric string.

Detailed description of Alphanumeric Dialing

This section provides a detailed description of the Alphanumeric Dialing feature.

With Alphanumeric Dialing, a user can place a data call with Data Call Setup and an alphanumeric string or alpha-name for the call-destination address. For example, a user can type 9+1-800-telefon instead of 9+1-800-835-3366 to place a call. Users need to remember only the alpha-name of the far-end terminating point.

With Alphanumeric Dialing, you can 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 name that the users enters is not in the table, the system denies the call attempt. The user receives either an *Invalid Address* message (DCP) or a *Wrong Address* message (ISDN-BRI).

Because data terminals use DCP or ISDN-BRI data modules to access the switch, dialing procedures vary:

- For DCP, at the **DIAL** prompt users type the alphanumeric name and then press **Enter**.
- For ISDN-BRI at the **CMD** prompt users type **d**, a space, the alphanumeric name and press **Enter**.

Hardware requirements for Alphanumeric Dialing

The Alphanumeric Dialing feature requires the following hardware:

- None

Administering Alphanumeric Dialing

This section describes:

- Any prerequisites for administering the Alphanumeric Dialing feature
- The screens that you use to administer the Alphanumeric Dialing feature
- Complete administration procedures for the Alphanumeric Dialing feature

You must complete the following actions before you can administer the Alphanumeric Dialing feature:

- None

Screens for administering Alphanumeric Dialing

Screen name	Purpose	Fields
<i>Alphanumeric Dial Table</i>	Map alphanumeric names to strings	• Alpha-name • Mapped String

Reports for Alphanumeric Dialing

The following reports provide information about the Alphanumeric Dialing feature:

- None

Considerations for Alphanumeric Dialing

This section provides information about how the Alphanumeric Dialing feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Alphanumeric Dialing under all conditions. The following considerations apply to Alphanumeric Dialing:

- You cannot use Alphanumeric Dialing on telephones that have Hayes modems.
- More than one alphanumeric name can refer to the same digit string.

Interactions for Alphanumeric Dialing

This section provides information about how the Alphanumeric Dialing feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Alphanumeric Dialing in any feature configuration.

- None

Announcements

Use the Announcements feature to administer announcements to play for people who call your business. For example, you can inform callers that the call cannot be completed as dialed, the call is in a queue, or that all lines are busy. An announcement is often used in conjunction with music.

Announcements can be *integrated* or *external*. Integrated announcements reside on a circuit pack in the carrier. External announcements are stored, and played back from adjunct equipment.

The Announcements feature supports the following capabilities:

- [VAL announcements](#)
- [VAL Manager](#)
- [Local announcements on the G700 Media Gateway](#)
- [Integrated announcements](#)
- [Barging-in on announcements](#)

Detailed description of Announcements

Use the Announcements feature to administer announcements for people to hear when they call in to your office. For example, you can inform callers that the call cannot be completed as dialed, that the call is in a queue, or that all lines are busy. An announcement is often used in conjunction with music.

Announcements can be integrated or external. Integrated announcements reside on a circuit pack in the carrier. External announcements are stored and played back from adjunct equipment.

For software release 11 or later of Avaya Communication Manager, multiple telephone sessions are allowed. One session is associated with each active integrated announcement circuit pack. Any announcement that is stored on a circuit pack can play through any port on the circuit pack. Any announcement, except those announcements that are 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.

The Announcements feature supports three general types of announcements:

- Delay announcement. Explains the reason for the delay and encourages the caller to wait.
- Forced announcement. Explains an emergency or service problem. Use when you anticipate numerous 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.

Use the Announcements feature for:

- A Vector Directory Number (VDN) of origin announcement.
- A security violation notification
- When:
 - DID calls cannot be completed as dialed.
 - Incoming private-network-access calls cannot be completed as dialed.
 - Calls enter a split or skill (first announcement).
 - DDC, UCD, or direct-agent calls are in queue for an assigned interval.
 - ACD and Call Vectoring calls are in queue for an assigned interval.
 - The destination of a call is a recorded announcement extension.
 - The system routes a call to a vector that contains an announcement step.
 - An announcement extension is specified as a coverage point.
 - An announcement is the incoming destination of a trunk group.
 - The Hospitality Automatic Wakeup feature is in use.

Announcement sources for the Avaya G700 Media Gateway

The Announcements feature provides an announcement source for each G700 Media Gateway that is registered to either an Avaya S8300 or an Avaya S8700 server. The S8700 supports 10 integrated TN750, TN2501, CWY1 announcement boards, plus an additional 250 G700 announcement sources, for a total of 260. The S8300 supports 50 G700 announcement sources.

NOTE:

The S8300 does not support standard port networks or TN-type boards. Also, the software resources for integrated boards and G700 sources are separated. The G700 announcement sources are counted separately towards the limit of 50 on the S8300, and 250 on the S8700.

VAL announcements

Avaya Voice Announcement over LAN (VAL) incorporates the TN2501AP, an integrated announcement circuit pack that:

- Plays announcements over the time-division multiplexing (TDM) bus, similar to the TN750C
- Has up to 1 hour of announcement storage time per circuit pack
- Has 33 ports (31 playback, 1 record, and 1 Ethernet)
- Supports a 10/100 MB Ethernet interface, to allow announcement and firmware file portability over a LAN, using FTP server functions
- Supports generated .wav announcement files

VAL Manager

Avaya VAL Manager is a stand-alone application that you can use to copy announcement files and Communication Manager announcement information to and from Communication Manager over a LAN connection. VAL Manager is part of the Avaya Integrated Management suite of products.

VAL Manager provides:

- Simplified administration for adding, changing, and removing Communication Manager announcement information
- The ability to back up and restore announcement files and information to and from Communication Manager
- The ability to view the status of announcements on the VAL circuit pack in any Communication Manager

Announcements can be stored in .wav files, which can be sent to a voice announcement over a LAN board without conversion. VAL Manager also provides a repository to back up and restore announcement files, and simplifies administration. With VAL Manager, you can view the current status of announcements, add, change, and remove announcements, and copy and back up announcement files from Avaya media servers to VAL Manager and back, over the LAN.

Local announcements on the G700 Media Gateway

G700 local announcements, which are also known as virtual voice announcements over LAN (or virtual VAL), provide twenty minutes total announcement time with fifteen playback channels. Use VAL Manager to manage local announcements on the G700 Media Gateway.

Integrated announcements

The Announcements feature allows you to administer either integrated announcements or announcements that are recorded on external devices. The external devices connect to the server by means of analog-line circuit packs or auxiliary trunk interfaces, such as a TN2183 or a TN763.

The system stores integrated announcements on a TN750 integrated-announcement circuit pack. The system can store multiple announcements on each circuit pack up to the system capacity. Each Integrated Announcement circuit pack has 16 ports, and can play up to 16 simultaneous announcements. The system can connect multiple users to each of these announcements.

Any announcement that is stored on a circuit pack can play through any port on the circuit pack. Any announcement, except those announcements that are administered for “barge-in,” can play simultaneously through multiple ports. All 16 ports can play the same announcement at the same time.

You must set the **Q** field to **y** on the **Announcements/Audio Sources** screen for each extension that you want to queue for integrated announcements. Calls to hear integrated announcements only queue when all 16 ports on the circuit pack that contains the announcement are busy. The same queuing pool is used for all boards. The system controls the announcement queue length for integrated announcements, but you must set the queue length for analog or aux-trunk announcements.

Single integrated announcement circuit packs

When your server has only one integrated announcement circuit pack, the circuit pack can be either a TN750, a TN750B, or a TN750C.

Multiple integrated announcement circuit packs

Multiple Integrated Announcement circuit packs can be installed in G3si and G3r servers. However, only one of these circuit packs can be a TN750 or a TN750B. Any additional circuit packs must be TN750C circuit packs.

**CAUTION:**

Do not copy announcements from a TN750C to a TN750 or TN750B. This action might corrupt the announcement data.

Backing up circuit packs

You must back up a TN750 or TN750B before someone:

- Removes a TN750 or TN750B from the server
- Shuts down power to the server

In both situations, the system loses any announcements that are stored on the circuit pack. Therefore, you must back up announcements that are stored on the TN750 or TN750B circuit packs to the Mass Storage System (MSS). When you insert or reset a circuit pack, or turn on the system, the system checks the circuit pack for announcements. If the system determines the circuit pack contains no announcements, the system automatically restores the announcements from the MSS.

**CAUTION:**

The announcements from the MSS that are automatically restored are the last announcements that were saved to the MSS. If multiple circuit packs are used, MSS might not contain the announcement for the B or the A circuit pack.

You do not need to back up a TN750C circuit pack. The TN750C has on-board FLASH memory backup. This on-board backup substantially reduces the time that is required for power-up restore, and eliminates the need for a manual save of the circuit pack contents. The system retains announcements on the TN750C circuit packs, even when someone removes the circuit pack, or when the system loses power. Therefore, the TN750C does not require the save-and-restore procedure. However, you can still use the save-and-restore procedure to copy the contents of a TN750C to another circuit pack.

Compression rates

The system stores integrated announcements on a TN750A at a compression rate of 32 Kbps. The system can store integrated announcements at one of three compression rates on the TN750B and the TN750C circuit packs. You administer the compression rate separately for each announcement extension. In this way, the system can store announcements with different compression rates on the same circuit pack. During playback, the server sets the port to the correct compression rate for the announcement that is playing.

- A 64-Kbps compression rate allows for 128 seconds of recorded announcement time per circuit pack.
- A 32-Kbps compression rate allows for 256 seconds of recorded announcement time per circuit pack. This rate is the default compression rate.
- A 16-Kbps compression rate allows for 512 seconds of recorded announcement time per circuit pack. The 16-Kbps rate does not provide a high-quality recording. This rate is adequate for VDN of Origin announcements, but Avaya does not recommend a 16-Kbps rate for customer announcements.

Barging-in on announcements

The system usually connects multiple callers to the beginning of an announcement, regardless of announcement type. However, you can also administer auxiliary trunk announcements, DS1 announcements, and integrated announcements to allow callers to begin listening to an announcement after the system starts to play the message. This capability is called “barge-in.”

When you administer barge-in, only one port plays the announcement at any one time. When the system routes a call to that announcement, the call immediately connects to the port, and the caller hears the announcement that is playing. Most administrators administer “barge-in” announcements to repeat continually while callers are connected to the port. In this way, the caller can listen until the system plays the entire announcement.

When barge-in is not enabled, and an announcement port is available when a call arrives, the system connects the call to the announcement. If no announcement port is available, and the announcement is administered with **no** as the queue option, the caller hears a busy signal or other feedback, depending on how the announcement was accessed.

If no announcement port is available and the announcement is administered with **yes** as the queue option, the call enters the announcement queue. When a port becomes available, the server connects the calls waiting in the queue to the beginning of the announcement. The system first connects the call that has been waiting in queue the longest and then connects as many calls as it can.

Hardware requirements for Announcements

The Announcements feature requires the following hardware:

- For integrated announcements, a TN2501AP or a G700 VVAL announcement circuit pack must be properly installed and configured.
- For external announcements, adjunct devices must be properly installed and configured.

Administering Announcements

The following steps are part of the administration process for the Announcements feature:

- [Displaying all announcements](#)
- [Adding announcement data modules](#)
- [Adding announcement extensions](#)
- [Recording and changing announcements](#)
- [Saving announcements](#)
- [Copying announcements](#)
- [Restoring announcements](#)
- [Deleting and erasing announcements](#)
- [Setting up continuous-play announcements](#)
- [Recording VAL announcements](#)
- [Converting announcement files to VAL format](#)
- [Using FTP to Manage VAL announcements](#)
- [Recording announcements for TTY callers](#)

This section describes:

- Any prerequisites for administering the Announcements feature
- The screens that you use to administer the Announcements feature
- Complete administration procedures for the Announcements feature

Prerequisites for administering Announcements

You must complete the following actions before you can administer the Announcements feature:

- None

Screens for administering Announcements

Screen name	Purpose	Fields
<i>Announcements/Audio Sources</i> (including Integrated Announcement Translations)	Add, change, or delete individual announcements and the properties of the announcements.	All
<i>Feature Access Code (FAC)</i>	Set up a feature access code (FAC) to use to activate the Announcements feature.	Announcement Access Code

Screen name	Purpose	Fields
Station	Set the Class of Service (COS) for using the Announcements feature.	COS
Data Modules — Netcon Data Module — System Port Data Module (SAP) — Announcement Data module	Use this screen to administer Save/Restore/Copy parameters.	All
Circuit Packs	Administer this screen if you administer the Board Location on the Announcements/Audio Sources screen or the Data Module screen, and do not have the circuit pack plugged in.	All
Feature-Related System Parameters	Administer this screen if you plan to use Announcements with the feature associated with each field shown in the Fields column at right.	• DID/Tie/ISDN Intercept Treatment • Controlled Outward Restriction Intercept Treatment • Controlled Termination Restriction (Do Not Disturb) • Controlled Station-to-Station Restriction
Hospitality	Administer this screen if you plan to use Announcements with the with the Hospitality feature.	• Announcement Type • Length of Time to Remain Connected to Announcement
Trunk Group	Administer this screen if you plan to use Announcements with the Hospitality feature.	Incoming Destination

Screen name	Purpose	Fields
<i>Coverage Path</i>	Administer this screen if you plan to use Announcements with the Call Coverage feature.	Coverage Points
<i>Hunt Group</i>	Administer this screen if you plan to use Announcements with the Hunt Groups feature.	• First Announcement Extension • Second Announcement Extension
<i>Call Vector</i>	Administer this screen if you plan to use Announcements with the Call Vector feature.	All fields that require announcements

Displaying all announcements

To display all existing administered announcement boards:

- 1 Type **list integrated-annc-boards**. Press **Enter**.

The system displays the **Integrated Announcements** screen ([Figure 8, Integrated Announcements screen](#), on page 122).

Figure 8: Integrated Announcements screen

list integrated-annc-boards					
INTEGRATED ANNOUNCEMENTS					
Board Location: 08A05					
Annc. Number	Internal Number	Announcement Extension	Name	Rate	Length in Seconds
1:	NA	658001	Collect_1_digit	64	4
2:	NA	658002	No_digits_collected_Goodbye	64	5
3:	NA	658003	Collect_5_digits	64	5
4:	NA	658004	voa1	64	6
5:	NA	658005	voa2	64	6

Adding announcement data modules

To add an announcement data module:

- 1 Type **add data-module next**. Press **Enter**.
The system displays the screen [\(Figure 9, Data Modules screen, on page 123\)](#).

Figure 9: Data Modules screen

DATA MODULE

Data Extension: 2002
Type: announcement
Board: 01B18
ITC: restricted__

Name: announcement data module
COS: 1
COR: 1
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**. Press **Enter**.
The Port field automatically changes to Board.
- 4 In the Board field, type the address of the announcement circuit pack. In this example, the address is **01B18**.
- 5 Press **Enter** to save your changes.

Adding announcement extensions

To add an announcement extensions:

- 1 Type **change announcements**. Press **Enter**.
The system displays the *Data Modules* screen [\(Figure 9, Data Modules screen, on page 123\)](#).

Figure 10: Announcements/Audio Sources screen

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 the extension you want to assign to this announcement.
- 3 In the **Type** field, enter the announcement type.
- 4 In the **Name** field, type a name for this announcement.

NOTE:

If you use the TN2501AP or the G700 Media Gateway VVAL, then you must enter a name in the **Name** field. This name becomes the name for the announcement file.

- 5 In the **Q** field, perform one of the following actions:
 - If you want calls to wait to hear an announcement if all the ports on the announcement circuit pack are busy, type **y**.
 - If you do not want calls to wait to hear an announcement if all the ports on the announcement circuit pack are busy, type **n**. If you type **n** in this field, a queue is not created, and the caller hears a busy signal or other feedback, depending on how the announcement was accessed.
- 6 The system displays N/A in the **QLen** (queue length) field. You cannot change this field for integrated announcements. The length of the queue for integrated announcements is preset.
- 7 In the **Pro** (protected) field, perform one of the following actions:
 - If you want to allow users with console permissions to change the announcement, type **n**.
 - If you do not want to allow users with console permissions to change the announcement, type **y**. If you type **y** in this field, the announcement is protected and cannot be changed.

NOTE:

The **Pro** field appears only if the **Type** field is integrated.

- 8 In the **Port** field, type one of the following values:
 - The location of the TN2501AP or the TN750 announcement circuit pack. In this example, the location is **01B18**.
 - **ggv9** for G700 VVAL, where **gg** is the gateway number of the G700 Media Gateway.
- 9 Repeat steps 2 -8 with the correct information for each announcement you want to record.
- 10 Press **Enter** to save your changes.

NOTE:

These steps only create 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 a file transfer protocol (FTP) session.

- 11 To ensure that the announcement administration is correct, type **list integrated-annc-boards**. Press **Enter**. The system displays the Integrated Announcements screen ([Figure 8, Integrated Announcements screen](#), on page 122), where you can review announcement settings.

Recording and changing announcements

You can use a system telephone or an attendant console to record a new announcement for callers to hear when callers dial a specific extension. Use the same procedure to change an existing announcement.

Prerequisites

You must complete the following actions before you can record announcements:

- Ensure that you have a telephone or attendant console with a class of service (COS) that provides console permissions to record announcements. For more information on Class of Service, see the “Class of Service” feature.

To record or change an announcement:

- 1 Dial ***xx** from a telephone or console, where **xx** is the feature access code (FAC) that is administered for the announcement. For more information on FACs, see the “Feature Access Code” feature.
 - If you hear dial tone, continue with Step 2.
 - If you hear a fast busy signal, hang up. Redial the FAC and the extension every 45 seconds until you hear dial tone.
- 2 Dial the announcement extension.
- 3 When you hear dial tone, dial **1** to begin recording.
 - If you hear a beep or a stutter tone, begin recording. If the circuit pack memory becomes full while you are recording, the system drops your connection and does not retain the announcement.
 - If you hear intercept tone, hang up. Record your announcement on another extension that is assigned to a different circuit pack.
- 4 To end the recording:
 - If you are using a digital telephone, press the pound key (**#**). You hear a dial tone again, and you can continue your session.
 - If you are using an analog telephone, hang up. If your analog telephone is not connected through lineside DS1, the system records an electrical click at the end of the recording. You must redial the Announcement FAC to continue your session.
- 5 If you are using a digital telephone, and you want to listen to the announcement you just recorded, dial **2**. The recording plays back through the handset.
- 6 If you are not satisfied with the announcement, press
 - **1** to rerecord the announcement
 - **3** to delete the announcement and end the recording session
- 7 To listen to the announcement after you hang up, dial the announcement extension from any telephone or attendant console. The announcement plays through the handset. When you dial an announcement extension directly, for which an announcement has not been recorded or Footpad to the announcement board, you hear silence instead of a busy tone with the VAL type sources, that is, the TN2501 or the G700 Gateway. With TN750 boards, you hear a busy tone.

NOTE:

You must wait 15 seconds after you record an announcement before you can dial the extension to hear the announcement replay. During this 15-second window, you cannot record a new announcement, and no one can play this announcement. However, you can rerecord the announcement during this 15-second period. To rerecord the announcement, dial the Announcement FAC, dial the extension, and then press **1** before the 15-second timer expires.

Saving announcements

You can save and back up announcements from an announcement circuit pack to system memory. Your system memory is a tape, hard disk, or a memory card, depending on your system. Use this procedure primarily for announcement circuit packs without built-in memory. VAL and VVAL announcements are backed up internally to nonvolatile FLASH memory.

To save or back up announcements when you have only one announcement circuit pack:

- Type **save announcements**. Press **Enter**. The system saves the announcements that are on the circuit pack. This process can take up to 40 minutes. You cannot administer your system while the system is saving announcements.

To save or back up announcements when you have more than one announcement circuit pack:

- Type **save announcements from xxxxx**, where **xxxxx** is the address of the circuit pack for which you want to save announcements. For example, **save announcements from 01B18** saves announcements on the circuit pack that is located at address 01B18.

Copying announcements

You can copy recorded announcements between back-up disks and tapes when you want to store your recorded announcements in more than one place.

To copy announcements between your back-up disk and tape:

- 1 Type **copy announcements**. Press **Enter**.

NOTE:

This task only applies if you use a TN750-series announcement circuit pack.

If you use the TN2501AP (VAL) announcement circuit pack or VVAL announcements, see [Recording VAL announcements](#).

Restoring announcements

You can restore announcements from system memory to an announcement circuit pack. Your system memory is a tape, hard disk, or a memory card, depending on your system.

If you have a duplicated system, your system always restores the announcements that are located on the active processor.

To restore announcements from system memory to the integrated announcement circuit pack:

- 1 Type **restore announcements disk**. Press **Enter** to restore announcements.

To restore announcements from system memory to an announcement circuit pack that has built-in memory:

- 1 Type **restore announcements from xxxxx**, where **xxxxx** is the address of the circuit pack for which you want to restore announcements. For example, **restore announcements from 01B18** restores announcements on the circuit pack that is located at address 01B18.
- 2 Press **Enter** to restore announcements.

Deleting and erasing announcements

You can use a system telephone or an attendant console to delete recorded announcements from an announcement circuit pack that does not have stored memory. When you delete an announcement from the circuit pack, you do not delete the announcement from your system backup tape or hard disk.

You can also erase the announcements that are stored on an integrated announcement circuit pack that has stored memory.

NOTE:

The system denies any attempt to delete an announcement while the announcement is playing, being transferred, or being backed up to FLASH memory.

To delete or erase an announcement:

- 1 Dial ***xx**, from a telephone or an attendant console, where **xx** is the Announcement FAC administered for the announcement. For more information on FACs, see the “Feature Access Code” feature.
- 2 When you hear dial tone, dial **xxxx**, where **xxxx** is the announcement extension.
- 3 When you hear dial tone, 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 (the reorder tone), and the system does not delete the announcement.
- 4 Hang up the telephone.
- 5 To ensure that an announcement was deleted, dial **xxxx**, where **xxxx** is the extension of the deleted announcement. You hear a busy signal if the announcement was deleted.
- 6 Repeat Steps 1 through 5 for each announcement that you want to delete. You can delete only one announcement at a time.

You might also want to remove the announcement extension from the system. To remove the extension:

- 1 Type **change announcements**. Press **Enter**.
The system displays the **Announcements/Audio Sources** screen ([Figure 10, Announcements/Audio Sources screen](#), on page 123).
- 2 Delete the information in the **Ext** and **Type** fields.
- 3 Press **Enter** to save your changes.

Finally, to erase announcements on the announcement circuit pack:

- 1 Type **erase announcements xxxxx**, where **xxxxx** is the address of the circuit pack that contains the announcements that you want to erase.
The system displays a warning message that confirms the deletion.
- 2 Press **Enter** to erase the announcements on the circuit pack.

Setting up continuous-play announcements

You can set up announcements to continuously repeat while callers are connected to the announcement, so that a caller can listen 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 telephone to hear an announcement at a particular time, they use only one port. The message repeats on that port until the last guest goes off-hook and the message ends.

To set up a continuous-play announcement:

- 1 Type **change announcements**. Press **Enter**.
The system displays the Announcements/Audio Sources screen ([Figure 10, Announcements/Audio Sources screen](#), on page 123).
- 2 Type **b** in the **Q** field on the same line as the extension for the announcement.
- 3 Leave the name that is in the **Name** field, or enter a new description for the announcement.
- 4 Press **Enter** to save your changes.

Recording VAL announcements

You can record an announcement for callers to hear when the callers dial a specific extension, or as part of call vectoring. Use the same steps to change an existing announcement.

You can record announcements in several ways:

- At a professional recording studio
- At a computer
- At a system telephone

Prerequisites

You must complete the following actions before you can record VAL announcements:

- Ensure that the announcement administration is complete. You must assign a name before you can record an announcement. For more information, see [Adding announcement extensions](#).
- If you replace a TN750C announcement circuit pack with the TN2501AP:
 - Get a list and a description of the announcements that are stored on the TN750C circuit pack.
 - Rerecord the announcements on a computer or at a professional recording studio as .wav files (CCITT μ -Law or A-Law, 8-KHz, 8-bit mono), so that the files are ready to transfer to the new announcement circuit pack after the circuit pack is installed and administered.
- If you replace an old announcement circuit pack with the new TN2501AP circuit pack:
 - Remove previous announcement administration.
 - Record new announcements for the TN2501AP.
 - Rerecord any announcements that currently reside on the TN750 circuit packs that you are replacing. You cannot transfer or restore TN750 announcements from a FLASH card, tape, or optical disk to the TN2501AP.



CAUTION:

When you change a VAL announcement, the TN2501AP is activated to perform a FLASH auto-save backup 5 minutes later. If you try to record a new announcement during this backup process, the new recording fails, and a denial event is logged to the **Event Report** screen.

- Announcement file format requirements

To be compatible with the TN2501AP circuit pack and Communication Manager, announcement recordings must have the following parameters:

- CCITT A-Law or CCITT μ -Law companding format (do not use PCM)
- 8-KHz 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 that is specified on the **Location Parameters** screen.
- Filenames for the TN2501 (VAL) boards cannot contain blank spaces nor any of the following characters:
 - Period (.)
 - Comma (,)
 - Forward slash (/)
 - Colon (:)
 - Asterisk (*)
 - Question mark (?)
 - Less than (<)
 - Greater than (>)
 - Backward slash (\)

The filename itself is case sensitive, but the file extension .wav must be lowercase.

Announcements that are recorded in this format occupy 8-KB per second of file space for each second of recorded speech. For example, a 10-second announcement creates an 80-K .wav file.

To record a VAL announcement at a computer:

- 1 At the computer, open the application that you use to record .wav files.
- 2 Set the recording parameters.
- 3 Speak into a microphone that is connected to the computer and record the announcement.
- 4 Play the announcement back at the computer before you transfer the file to the VAL circuit pack.

To use a system telephone to record an announcement, see [Recording and changing announcements](#).

Converting announcement files to VAL format

If you share recordings in a multisite environment with Communication Manager and Avaya Interactive Voice Response systems, you must convert announcement files for use on either system. If you must convert an announcement file to the required format, you can use a sound recording utility application to do so.

To convert a previously recorded announcement, or a file that is compatible with an Avaya Interactive Voice Response system, to a format that is compatible with Communication Manager:

- 1** Open the sound recording application on your computer. For example, you might use Microsoft Windows Sound Recorder.
- 2** Open the file that you want to convert.
- 3** Check the file properties to determine if you must change the parameters.
- 4** If you must change the recording parameters, look for a conversion tool. Some tools have a **Convert Now** option. Other tools use **Save As**).
- 5** Change the file parameters to parameters that are 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, and change the default setting to match the required parameters if necessary. Note that 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 the Avaya Interactive Voice Response (IVR)

The Avaya Interactive Voice Response (IVR) has a recording conversion utility that supports file formats that are similar to the formats that VAL requires. However, the conversion utility can read only PCM-format announcement files.

To convert an announcement file for use on an Avaya Interactive Voice Response systems:

- 1** If the companding format of the file is already PCM, go to Step 5.
If you are not sure what the file format is, continue with Step 2.
- 2** At a computer, open the sound recording application.
- 3** Open the file that you want to convert.
- 4** Save (**Convert Now** or **Save As**) the announcement with these parameters:
 - Format: PCM
 - Bits/Sample: 8
 - Sample Rate: 8-KHz
 - Mono (channels = 1)

NOTE:

The recording conversion utility requires that announcement files be in PCM format. VAL files must be in CCIT A-Law or μ -Law format.

- 5** Open the file in the recording conversion utility.
- 6** Convert the file to SSP format.

Deleting VAL announcements

Prerequisites

You must complete the following actions before you can delete VAL announcements:

- Look up the announcement information. Type **list directory board**. Press **Enter**.
- Determine which announcements that you want to delete, either by extension or file name.
- Decide whether you are:
 - [Using the SAT to delete individual VAL announcement files](#)
 - [Using the SAT to delete all VAL announcements on a circuit pack](#)
 - [Deleting and erasing announcements](#)

NOTE:

The system denies any attempt to delete an announcement while the announcement is playing, being transferred, or backed up to FLASH memory.

Using the SAT to delete individual VAL announcement files

Use the SAT to delete an announcement using the System Administration Terminal (SAT):

- 1 Type **remove file board board-location /annc/filename.wav**, where **filename.wav** is the name of the file that you want to delete. Press **Enter**.

NOTE:

File names 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. The **/Closed.wav** portion indicates to delete the file **Closed.wav**.

Using the SAT to delete all VAL announcements on a circuit pack

Use the SAT to delete all announcement files on a VAL circuit pack:

- 1 Type **busyout board board-location**, where **board-location** is the 5-character board number assigned to the board. Press **Enter**.

NOTE:

When the VAL board is busied out,

- Both the RSCL and the Ethernet ports are busied out.
- The firmware takes down the Ethernet link.
- FTP is disabled because the Ethernet link is not functioning.
- Announcements on that circuit pack cannot play.

- 2 Type **erase announcements board board-location**, where **board-location** is the 5-character board number assigned to the board. Press **Enter**.



CAUTION:

This command deletes the specified announcement file in both RAM and FLASH memory. The board firmware ignores the protect flag (Pro field) when you erase the announcement files.

- 3 Type **list directory board**. Press **Enter**.
- 4 Verify that no files are listed.

NOTE:

The announcement directory on the TN2501AP or the G700 Media Gateway is **/annnc**.

- 5 Type **list integrated-annnc-boards**. Press **Enter**.
Check the list. The **Length in Seconds** field should show **0** if the announcement was deleted.

To use a system telephone to delete VAL announcements, see [Deleting and erasing announcements](#).

Using FTP to Manage VAL announcements

This section includes information on how to set up and terminate a file transfer protocol (FTP) session, and outlines announcement tasks that you can perform in an FTP session.

An FTP session has 3 basic components:

- [Setting up an FTP session](#)
- [Performing tasks in an FTP session](#)
- [Ending an FTP session](#)



SECURITY ALERT:

Read and observe all Security Alerts regarding enabling and disabling the VAL (TN2501AP or embedded G700 Gateway circuit packs) file system and FTP sessions into it.

Setting up an FTP session

To set up an FTP session into the VAL circuit pack, you must:

- Prepare the VAL circuit pack for the FTP session. This action:
 - Allows an FTP session on an individual VAL circuit pack
 - Creates an ftp-login and ftp-password for that session
- Start an FTP session from a computer or a network management terminal. Before you can start the FTP session, you must know the:
 - IP address of the VAL circuit pack.
 - ftp-login and ftp-password of the VAL circuit pack.

Preparing the VAL circuit pack for the FTP session

To prepare the VAL circuit pack for the FTP session, including setting the user name and the password:

- 1 Type **enable filesystem board board-location login ftp-username ftp-password**, where **ftp-username** is a string of 3 to 6 alphanumeric characters, and **ftp-password** is a string of 7 to 11 alphanumeric characters. Press **Enter**.

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 to 6 characters) for this session is **romeo**, and the ftp-password (7 to 11 characters) is **shakespeare**.

When the FTP session on the circuit pack is enabled, the announcement and the firmware files are available to anyone who knows the IP address of the VAL circuit pack, the ftp-username, and the ftp-password.



SECURITY ALERT:

Avaya recommends that you use a unique ftp-login and ftp-password for each 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, file names 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 another 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.

Starting an FTP session

To start an FTP session:

- 1 At the FTP client, type **ftp val-ip-address**, where **val-ip-address** matches the VAL IP address administered on the **IP Node Names** screen. Press **Enter**.
- 2 At the username prompt, type the ftp-username. Press **Enter**.
- 3 At the password prompt, type the ftp-password. Press **Enter**.

The system displays the message `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. To set the system to binary mode, at the FTP client, type **bin**. Press **Enter**.

The system sets the **Types** field to **I**, binary mode.



CAUTION:

If you do not transfer announcement files in binary mode, the files can be corrupted, and the FTP session can fail.

Performing tasks in an FTP session

Once the FTP session is established, you can perform any of these announcement tasks:

- [Moving announcements from the VAL circuit pack](#)
- [Using FTP to delete VAL announcements](#)
- [Moving announcements to a VAL circuit pack or another LAN device](#)
- [Combining copy and move 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 that you copy the file from the VAL circuit pack to the default directory of the FTP client.

To back up or save an announcement from the VAL board to the client computer through an FTP session:

- 1 At the FTP client, type **get filename.wav**, where **filename.wav** is the name of the announcement file. Press **Enter**.

The system writes the announcement file to the directory from which you started the FTP session.

NOTE:

FTP upload or download of announcement files does not preserve the created time stamp. The file receives the current date and time when the file is written to the circuit pack or to the computer.

- 2 List the contents of the FTP client directory and ensure that the announcement file is listed.
- 3 End the FTP session. See [Ending an FTP session](#) for the procedure.

Using FTP to delete VAL announcements

You can use FTP to 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 memory.

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**, where **filename.wav** is the name of the announcement file. Press **Enter**.

NOTE:

The announcement file is removed only from volatile RAM memory immediately. 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 is visible when you view the announcement directory from the FTP client.

- 4 End the FTP session. See [Ending an FTP session](#) for the procedure.
- 5 At the SAT, type **change announcements**. Press **Enter**.
The system displays the **Announcements/Audio Sources** screen ([Figure 10, Announcements/Audio Sources screen](#), on page 123).
- 6 Delete the entire line associated with the announcement.
- 7 Press **Enter** to save your changes.

Moving announcements to a VAL circuit pack or another LAN device

You can copy an announcement file on the VAL circuit pack to another LAN device. Ensure that you *have not just* administered the announcement on the **Announcements/Audio Sources** screen. If announcement administration precedes the file transfer:

- The announcement appears with a zero (0) length on the **Integrated Announcements** screen.
- The Time Remaining field on the **Integrated Announcements** screen does not refresh to reflect the presence of the new announcement file on the circuit pack.

Use the following procedure to ensure that the announcement length is accurate:

- 1 Administer the announcement on Communication Manager with the **change announcements** command. Use the identical file name 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 system 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. Again, the system returns a busy signal at the first play attempt.
- 4 Rerecord this announcement with the same filename at a system telephone. See [Recording and changing announcements](#) for the procedure.

To copy an announcement to a VAL circuit pack or another LAN device:

- 1 At the FTP client, type **put filename.wav**, where **filename.wav** is the name of the announcement file. Press **Enter**.
- 2 List the contents of the VAL announcement directory or the LAN device, and ensure that the announcement file is listed.

NOTE:

FTP upload or download of announcement files does not preserve the created time stamp. The file receives the current date and time when the file is written to the circuit pack or to a computer.

- 3 After you ensure that the announcement is on the VAL circuit pack, administer the announcement on Communication Manager with the **change announcements** command. Use the identical file name in the **Name** field *without spaces or the .wav file extension*.
- 4 End the FTP session. See [Ending an FTP session](#) for the procedure.

Combining copy and move tasks

When you combine copy (the **get** command) and move (the **put** command) announcement files, you can rearrange VAL announcements.

To move an announcement to a VAL circuit pack from another VAL circuit pack in an FTP session:

NOTE:

You must first establish an FTP session into the circuit pack *from which you are restoring an announcement*.

- 1 List the directory contents of the circuit pack containing the announcement you want to copy, and ensure that the announcement file is listed.
- 2 At the FTP client, type **get filename.wav**, where **filename.wav** is the name of the announcement file. Press **Enter**.
The system writes a copy of the file to the directory from which you started the FTP session.
- 3 List the contents of the FTP client directory, and ensure that the announcement is listed.
- 4 End the FTP session to the circuit pack *from which you copied the announcement file*. See [Ending an FTP session](#) for the procedure.
- 5 Set up a new FTP session into the destination VAL circuit pack. See [Setting up an FTP session](#) for the procedure.
- 6 At the FTP client, type **put filename.wav**, where **filename.wav** is the name of the announcement file. Press **Enter**.
- 7 List the contents of the VAL announcement directory, and ensure that the announcement is listed.
- 8 End the FTP session to the circuit pack *to which you copied the announcement file*. See [Ending an FTP session](#) for the procedure.

Ending an FTP session

FTP sessions to a VAL circuit pack originate at the FTP client end.

To end an FTP session, you can:

- 1 Type **bye** or **quit** to log out from the FTP client. Press **Enter**
- 2 Type **disable filesystem board *board-location***, where ***board-location*** is the 5-character board number assigned to the board. Press **Enter**. This command clears the ftp-username and the ftp-password.
 - Allow the system to time out after 10 minutes of inactivity.



SECURITY ALERT:

A higher degree of system security is provided if you both log out of the FTP session and disable the VAL circuit pack.

NOTE:

If you only disable the circuit pack file system, you can continue your FTP session. However, new FTP session logins are not allowed.

Recording announcements for TTY callers

Record announcements for Teletypewriter device (TTY) callers in the same way as you record voice announcements. However, instead of recording from the handset of your telephone, you record from a TTY device. You can use an acoustic coupler into which you place the telephone handset to attach the device to your telephone, or plug the TTY device directly into the back of a digital telephone. After you call the announcement extension, and press **1** to record, you type the announcement into the TTY device.

If you use an acoustic coupler to connect your telephone for recording, you can record TTY and voice into a single announcement. In this case, after you press **1** to record, you can type the TTY message, and then immediately pick up the handset to record the voice message. For this type of recording, digital telephones also offer the option to press the pound key (**#**) to complete the recording, which eliminates any extraneous noise at the end of the recording. Unfortunately, this method for combined TTY and voice recordings is likely to create extraneous noise in the middle of your announcements.

A better alternative to recording with your telephone is to create .wav files on other recording applications, and then copy and save the .wav files to your announcement board. See [Moving announcements to a VAL circuit pack or another LAN device](#) for the procedure. In this case, the announcement files must meet the same criteria as voice recordings. See [Announcement file format requirements](#) for more information.

Reports for Announcements

The following reports provide information about the Announcements feature:

- Denial Events Log

If a working VAL announcement file is deleted over FTP, the next attempt to play the announcement fails, and the system adds a software event to the Denial Events Log. You can view the Denial Events log to see if the announcement was deleted, and to see if other events occurred that are related to announcements. To access this report, type **display events** and select **denial** in the Category field.

- Announcement Measurements

You can view a report of announcement measurements. This report includes how many times an announcement was queued to play, how many callers dropped while in the queue, and how many times all announcement ports were busy during the report period.

For detailed information on these reports and the associated commands, see *Reports for Avaya Communication Manager*.

Considerations for Announcements

This section provides information about how the Announcements feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Announcements under all conditions. The following considerations apply to Announcements:

- Announcements that are recorded on announcement circuit packs without built-in memory are lost if the announcements are not saved to system memory before power is shut down, or the circuit pack is reset or removed.

Interactions for Announcements

This section provides information about how the Announcements feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Announcements in any feature configuration.

- **Automatic Call Distribution (ACD)**

Recorded announcements are used extensively for ACD, Call Vectoring, Call Prompting, Expert Agent Selection (EAS), Vector Directory Number (VDN) of Origin Announcement, Direct Department Calling, and Uniform Call distribution (UCD) features. See the individual features for interaction details.

- **Automatic Wakeup**

Recorded Announcement allows Automatic Wakeup to use the built-in TN750B or later-suffix announcement circuit pack, in place of the 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 go off-hook to receive an announcement at a particular time, the guests use only one port. The message repeats on the port until the last guest goes off-hook and the message ends.

Troubleshooting announcements

This section lists the known or common problems that users might experience with the Announcements feature.

Problem	Possible cause	Action
Poor sound quality.	Bad file format.	Ensure that the file formats are compatible. A good announcement file format must have the following parameters: • 8-Kbps sample rate • 8-bit resolution (bits per sample) • A-law or Mu-law companding format • Mono (channels = 1)
	Incorrect companding mode.	Ensure that the same companding mode is administered on page 1 of the Location Parameters screen (type change location-parameters) for both environments.
An announcement does not play.	If a working VAL announcement file is deleted over FTP, the next attempt to play the announcement fails, and the system adds a software event to the Denial Events Log.	View the Denial Events log to determine if the announcement was deleted. If the announcement was deleted, restore or rerecord the announcement.
The system displays error code E28 when you attempt to restore announcements.	An integrated announcement circuit pack is not installed in the system.	
The system displays error code E31 when you attempt to restore announcements.	A call is connected to the announcement on the circuit pack and the port is busy..	Wait for the call to disconnect, and try to restore announcements again.

Attendant Auto-Manual Splitting

Use the Attendant Auto-Manual Splitting feature to allow the attendant to announce an incoming call to a user without being heard by the calling party. The attendant can also use the Attendant Auto-Manual Splitting feature to consult privately with the called party without being overheard.

Detailed description of Attendant Auto-Manual Splitting

This section provides a detailed description of the Attendant Auto-Manual Splitting feature.

The Attendant Auto-Manual Splitting feature allows the attendant to split the calling party away from a conversation. The attendant can then confidentially determine if the called party wants to accept the call.

The system automatically activates the Attendant Auto-Manual Splitting feature when the attendant, who is active on a call, presses any of the following buttons:

- Start
- An extension
- Any Hundreds Select button plus an extension
- Trunk Group Select

Hardware requirements for Attendant Auto-Manual Splitting

The Attendant Auto-Manual Splitting feature requires the following hardware:

- Attendant console

Administering Attendant Auto-Manual Splitting

The following steps are part of the administration process for the Attendant Auto-Manual Splitting feature:

- The system activates the Attendant Auto-Manual Splitting feature when you set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

This section describes:

- Any prerequisites for administering the Attendant Auto-Manual Splitting feature
- The screens that you use to administer the Attendant Auto-Manual Splitting feature
- Complete administration procedures for the Attendant Auto-Manual Splitting feature

Prerequisites for administering Attendant Auto-Manual Splitting

You must complete the following actions before you can administer the Attendant Auto-Manual Splitting feature:

- None

Screens for administering Attendant Auto-Manual Splitting

Screen name	Purpose	Fields
<i>Attendant Console</i>	Set up an attendant console.	All

Reports for Attendant Auto-Manual Splitting

The following reports provide information about the Attendant Auto-Manual Splitting feature:

- None

Considerations for Attendant Auto-Manual Splitting

This section provides information about how the Attendant Auto-Manual Splitting feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Auto-Manual Splitting under all conditions. The following considerations apply to Attendant Auto-Manual Splitting:

- None

Interactions for Attendant Auto-Manual Splitting

This section provides information about how the Attendant Auto-Manual Splitting feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Auto-Manual Splitting in any feature configuration.

- None

Attendant Auto Start and Don't Split

Use the Attendant Auto Start and Don't Split features to allow an attendant to press any key on the keypad to start a call without the need to first press the Start button.

Detailed description of Attendant Auto Start and Don't Split

This section provides a detailed description of the Attendant Auto Start and Don't Split features.

Auto Start and Don't Split are two features that work together. These two features reduce the number of buttons that attendants must press to handle calls.

Auto Start

If the attendant is on a call and presses any key on the keypad, the system automatically splits the current call, places the call on hold, and dials the next call. The system disables the Start button, and does not allow end-to-end signaling.

To extend a current call to another extension, the attendant dials the extension. The system automatically places the current call on hold. Once the called party answers, the attendant presses the Release button to extend the call.

Don't Split

To deactivate the Auto Start feature and not split a current call, the attendant presses the Don't Split button. One use of the Don't Split feature is to send touchtones to pick up answering machine messages. When the Don't Split feature is active, parties on the call hear the keys that the attendant presses.

To reactivate the Auto Start feature and allow end-to-end signaling, the attendant can:

- Press the Don't Split button again
- Press Cancel
- Allow the current call to terminate

Hardware requirements for Attendant Auto Start and Don't Split

The Attendant Auto Start and Don't Split features require the following hardware:

- An attendant console

Administering Attendant Auto Start and Don't Split

The following steps are part of the administration process for the Attendant Auto Start and Don't Split features:

- [Assigning a Don't Split button](#)

This section describes:

- Any prerequisites for administering the Attendant Auto Start and Don't Split features
- The screens that you use to administer the Attendant Auto Start and Don't Split features
- Complete administration procedures for the Attendant Auto Start and Don't Split features

Prerequisites for administering Attendant Auto Start and Don't Split

You must complete the following actions before you can administer the Attendant Auto Start and Don't Split features:

- Set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Auto Start and Don't Split

Screen name	Purpose	Fields
<i>Attendant Console</i>	Assign a Don't Split button.	Any unassigned button in the Feature Button Assignments area

Assigning a Don't Split button

To assign a **Don't Split** button:

- 1 Type **change attendant *n***, where *n* is the number of the attendant console. Press **Enter**.
The system displays the *Attendant Console* screen.
- 2 Press **Next** until you see the Feature Button Assignments area ([Figure 11, Attendant Console screen](#), on page 145).

Figure 11: Attendant Console screen

change attendant 1		ATTENDANT CONSOLE		Page 3 of 4
FEATURE BUTTON ASSIGNMENTS				
1: split		13: _____		
2: dont-split		14: _____		
3: act-tr-grp		15: _____		
4: deact-tr-g		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__ *		

- 3 In the Feature Button Assignments area, assign **dont-split** to an available button.
In this example, we assign **dont-split** to button 2.
- 4 Press **Enter** to save your changes.

For more information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Reports for Attendant Auto Start and Don't Split

The following reports provide information about the Attendant Auto Start and Don't Split features:

- None

Considerations for Attendant Auto Start and Don't Split

This section provides information about how the Attendant Auto Start and Don't Split features behave in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Auto Start and Don't Split under all conditions. The following considerations apply to Attendant Auto Start and Don't Split:

- If an attendant enables Auto Start, and then dials an Automatic Alternate Routing (AAR) number where the values in the **min** and the **max** fields in the **AAR Analysis Table** are not equal, the attendant must press the pound key (#) after the digit string. If the attendant does not press the pound key, the system does not process the call.

Interactions for Attendant Auto Start and Don't Split

This section provides information about how the Attendant Auto Start and Don't Split features interact with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Auto Start and Don't Split in any feature configuration.

- Call Detail Recording (CDR) Account Code Dialing

If the system is using Call Detail Recording Account Code Dialing, the Auto Start and the Don't Split features are not activated.

- Visually Impaired Service (VIS)

If VIS is activated or deactivated while Don't Split is active, the system automatically deactivates the Don't Split feature.

Attendant Backup

Use the Attendant Backup feature to give access to most attendant console features from specially administered backup telephones.

Detailed description of Attendant Backup

This section provides a detailed description of the Attendant Backup feature.

You can configure your system to have backup telephones for the attendant. With the Attendant Backup feature, designated users can answer calls that the attendant cannot immediately answer. These designated users can also provide some of the services that the attendant usually provides.

Designated users can access some attendant console features from the backup telephones when the:

- Queue of received calls reaches the administered warning levels
 - Number of calls in the queue
 - Time that a particular call has been in the queue
- Attendant console is in Night Service

A user with console permissions can:

- Activate Automatic Wakeup for another extension
- Activate and deactivate controlled restrictions for another extension or a group of extensions
- Activate and deactivate Do Not Disturb for another extension or a group of extensions
- Activate Call Forwarding for another extension
- Add and remove agent skills
- Record integrated announcements

To assign the Attendant Backup capabilities, you use **Console Permissions** field on the **Class of Service** screen.

You can assign the Attendant Backup feature to only multiappearance telephones. Avaya recommends the following multiappearance telephone models as a backup for the attendant:

- 6408
- 6416
- 6424

Attendant Backup supports the following capabilities:

- Attendant Backup Alerting

Attendant Backup Alerting

The Attendant Backup Alerting feature notifies the backup telephones that the attendant needs assistance to handle calls. The system provides both audible and visual alerting to backup telephones when the calls in the attendant queue reaches the administered warning levels. When the queue drops below the administered warning level, alerting stops.

Audible alerting also occurs when the attendant console is in Night Service, regardless of the size of the attendant queue.

Once the system alerts the backup telephones, designated users can answer an attendant call by:

- Pressing the **atd-qcalls** button
- Dialing the feature access code (FAC) for Trunk Answer Any Station (TAAS) feature.

Hardware requirements for Attendant Backup

The Attendant Backup feature requires the following hardware:

- An attendant console
- Multiappearance telephones on your system

Administering Attendant Backup

The following steps are part of the administration process for the Attendant Backup feature:

- [Setting up Attendant Backup telephones](#)

This section describes:

- Any prerequisites for administering the Attendant Backup feature
- The screens that you use to administer the Attendant Backup feature
- Complete administration procedures for the Attendant Backup feature

Prerequisites for administering Attendant Backup

You must complete the following actions before you can administer the Attendant Backup feature:

- Assign a Class of Service (COS) value to the multiappearance telephones that you use for Attendant Backup. On the **Class of Service** screen, you must set the **Client Room** field to **n**, and the **Console Permissions** field to **y**.

For more information, see [Considerations for Attendant Backup](#) on page 154 and the “Class of Service” feature.

- On the **Feature Access Code (FAC)** screen, type a FAC in the Trunk Answer Any Station field. Then share the FAC with each of the attendant backup users. For more information, see the “Feature Access Code” feature.
- You also need to set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator’s Guide for Avaya Communication Manager*.

Screens for administering Attendant Backup

Screen name	Purpose	Fields
Class of Service	Ensure that the Attendant Backup feature is set up correctly.	Client Room
	Assign console permissions to the backup telephones.	Console Permissions
Console Parameters	Administer the Attendant Backup Alerting feature.	Backup Alerting
	Set the queue warning parameters.	• Calls in Queue Warning • Time in Queue Warning (sec)
Feature Access Code (FAC)	Assign a FAC for backup telephone users.	Trunk Answer Any Station
Station	Assign an atd-qcalls feature button for backup telephone users.	Any available button field in the Feature Button Assignments area.

Setting up Attendant Backup telephones

To allow your system to alert the backup telephones, perform the following procedures:

- [Configuring your system](#)
- [Defining Class of Service console permissions](#)
- [Assigning console permissions to backup telephones](#)
- [Training designated users](#)

Configuring your system

To configure the system for the Attendant Backup feature:

- 1 Type **change console-parameters**. Press **Enter**.

The system displays the **Console Parameters** screen ([Figure 12, Console Parameters screen](#), on page 150).

Figure 12: Console Parameters screen

change console-parameters	CONSOLE PARAMETERS	Page 1 of 5
Attendant Group Name: OPERATOR		
COS: 0	COR: 0	
Calls in Queue Warning: 5	Attendant Lockout? y	
Ext Alert Port (TAAS):		
CAS: none		
IAS (Branch)? n	Night Service Act. Ext.:	
IAS Att. Access Code:	IAS Tie Trunk Group No.:	
Backup Alerting? y	Alternate FRL Station:	
Attendant Vectoring VDN:	DID-LDN Only to LDN Night Ext? n	

- 2 In the Backup Alerting field, type y.

The system can now notify any telephone that has an **atd-qcalls** feature button when:

- the attendant queue reaches the warning level
- the console is in night service

- 3 In the Calls in Queue Warning field, type the maximum number of calls that can be in the attendant queue before the system alerts the backup telephones.

In this example, we limit the number to five calls.

- 4 Press **Next** until you see the Time in Queue Warning (sec) field ([Figure 13, Console Parameters screen](#), on page 150).

Figure 13: Console Parameters screen

change console-parameters	CONSOLE PARAMETERS	Page 2 of 5
TIMING		
Time Reminder on Hold (sec): 10	Return Call Timeout (sec): 10	
Time in Queue Warning (sec): 120		
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:	

- 5 In the **Time in Queue Warning (sec)** field, type the maximum number of seconds that a caller can be in the attendant queue before the system alerts the backup telephones.
In this example, we assign a limit of 120 seconds.
- 6 Press **Enter** to save your changes.

Defining Class of Service console permissions

To define Class of Service (COS) console permissions for the backup telephones:

- 1 Type **change cos**. Press **Enter**.
The system displays the **Class of Service** screen ([Figure 14, Class of Service screen](#), on page 151).

Figure 14: Class of Service screen

change cos	CLASS OF SERVICE																Page 1 of 1
	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	y	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	y	n	n	n	n	n	n	n	n	
Personal Station Access (PSA)	n	n	n	n	n	n	n	y	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	y	n	n	n	n	n	n	n	n	n	
Trk-to-Trk Restriction Override	n	n	n	n	n	n	n	y	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	

- 2 In the **Console Permissions** row, change **n** to **y** in a numbered column.
Each column is a COS identification number. There are 16 available classes of service. In this example, we assign **Console Permissions** to COS 7.
- 3 To assign any other permissions for this COS, change **n** to **y** in the corresponding column.
- 4 Press **Enter** to save your changes.

Assigning console permissions to backup telephones

To assign console permissions to backup telephones:

- 1 Type **change station n**, where **n** is the extension of the backup telephone. Press **Enter**.
In this example, we change extension 4345. The system displays the **Station** screen ([Figure 15, Station screen](#), on page 152).

Figure 15: Station screen

change station 4345		Page 1 of 5
STATION		
Extension: 4345	Lock Messages? n	BCC: 0
Type:	Security Code:	TN: 1
Port:	Coverage Path 1:	COR: 1
Name:	Coverage Path 2:	COS: 7
	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?	
Model:		
	Media Complex Ext:	
	IP Softphone? y	

- 2 In the COS field, assign the COS that you created for the console permissions.
In this example, we assign COS 7 to this telephone.
- 3 Press **Next** until you see the Feature Button Assignments area ([Figure 16, Station screen](#), on page 152).

Figure 16: Station screen

change station 4345		Page 3 of 5
STATION		
FEATURE BUTTON ASSIGNMENTS		
9: crss-alert		
10: atd-qcalls		
11:		
12:		
13:		
14:		
15:		
16:		

- 4 In the Feature Button Assignments area, assign **atd-qcalls** to an available button.
In this example, we assigned **atd-qcalls** to button 10. The **atd-qcalls** button provides visual alerting for this telephone. When the **atd-qcalls** button:
 - Is dark, there are no calls in the attendant queue.
 - Shows a steady light, there are calls in the attendant queue.
 - Shows a flashing light, the number of calls in the attendant queue exceeds the limit that you set for the queue. At this point, the backup telephone user also hears an alert signal every 10 seconds.
- 5 Press **Enter** to save your changes.
- 6 Repeat this process for each backup telephone.

Training designated users

Finally, you must train the backup users. The backup users must know:

- How to interpret the **atd-qcalls** button lights
- The FAC for the TAAS feature

This step is for users whose telephones do not have an **atd-qcalls** button,

- Your procedure how to answer attendant calls
- Your procedure how to transfer attendant calls

End-user procedures for Attendant Backup

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Attendant Backup:

- If the **atd-qcalls** button on your telephone is flashing, the number of calls in the attendant queue exceeds the maximum limit. You also hear an alert signal every 10 seconds. To answer the call:
 - Press the **atd-qcalls** button on your telephone
 - Dial the FAC for the TASS feature

Reports for Attendant Backup

The following reports provide information about the Attendant Backup feature:

- None

Considerations for Attendant Backup

This section provides information about how the Attendant Backup feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Backup under all conditions. The following considerations apply to Attendant Backup:

- Backup telephone users must meet the following criteria to answer alerting calls in the attendant queue:
 - Have a multiappearance telephone
 - Have an **atd-qcalls** feature button assigned to the telephone
 - Be assigned a Class of Service (COS) that has the **Client Room COS** field set to **n**. Classes of Service are defined on the **Class of Service** screen.

If the **Client Room COS** field is set to **y**, the user receives intercept treatment when the user tries to use the Attendant Backup feature.
- When the attendant console is in day mode and the Attendant Backup feature is disabled, backup telephone users do not hear the audible alert signal. Backup telephone users also cannot dial the FAC for TASS to answer attendant calls.

Interactions for Attendant Backup

This section provides information about how the Attendant Backup feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Backup in any feature configuration.

- **Ringer Cutoff**

Activating the Ringer Cutoff feature disables the audible alerting signal. If a backup telephone has Ringer Cutoff activated, the system audibly alerts the telephone only when the attendant queue exceeds the queue warning level.

If Ringer Cutoff is not activated, the system audibly alerts the backup telephone every 10 seconds until the attendant queue falls below the queue warning level.
- **Trunk Answer Any Station (TAAS)**

If the system is in night mode and a TAAS port is not assigned, backup telephone users must dial the FAC for TASS to answer queued calls.
- **Tenant Partitioning**

You cannot use the Attendant Backup feature if Tenant Partitioning is enabled.

Attendant Call Waiting

Use the Attendant Call Waiting feature to allow an attendant to notify a single-line telephone user, who is on an existing call, that another call is waiting. Once the attendant notifies the user, the attendant can answer other calls.

Detailed description of Attendant Call Waiting

This section provides a detailed description of the Attendant Call Waiting feature.

The system automatically activates the Attendant Call Waiting feature whenever an attendant originates or extends a call to a single-line telephone user who is on an existing call. When the system activates Attendant Call Waiting, the:

- Attendant hears a call waiting ringback tone
- Telephone user hears a call waiting tone
- Calling party does not hear a tone

The attendant can place a call in progress on hold. After the attendant answers the held call, the attendant can use the Hold button to return to the held call. The attendant can alternate between the two calls.

For example, assume that extension 123, a single-line telephone, is busy. An attendant extends an incoming call to extension 123. The attendant hears the call waiting ringback tone, which indicates that Attendant Call Waiting is activated. The attendant can:

- Announce the call-waiting condition to the calling party
- Cancel the Attendant Call Waiting feature, and ask the calling party to call again later
- Release the caller, or place the call on hold at the console

Releasing an attendant-originated call drops the call completely.

The user at extension 123 hears a call-waiting tone, and knows that a call is waiting. The user can:

- Terminate the existing call
- Place the existing call on hold, and answer the waiting call

If the user does not answer the waiting call before a preassigned time interval, the system returns the call to the attendant.

Hardware requirements for Attendant Call Waiting

The Attendant Call Waiting feature requires the following hardware:

- An attendant console
- A single-line telephone

Administering Attendant Call Waiting

The following steps are part of the administration process for the Attendant Call Waiting feature:

- [Setting up single-line telephones](#)
- [Changing the call-waiting signal](#)
- [Modifying timed intervals](#)

This section describes:

- Any prerequisites for administering the Attendant Call Waiting feature
- The screens that you use to administer the Attendant Call Waiting feature
- Complete administration procedures for the Attendant Call Waiting feature

Prerequisites for administering Attendant Call Waiting

You must complete the following actions before you can administer the Attendant Call Waiting feature:

- The system activates the Attendant Call Waiting feature when you set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Call Waiting

Screen name	Purpose	Fields
Station	Allow the attendant to send calls to busy single-line telephones.	Att. Call Waiting Indication
Feature-Related System Parameters	Change the call-waiting signal.	Attendant Originated Calls
Console Parameters	Indicate the maximum number of seconds that a call can remain on hold before the system alerts the attendant.	Timed Reminder on Hold
	Indicate the maximum number of seconds that a call can remain on hold before the call returns to the attendant.	Return Call Timeout

Setting up single-line telephones

To allow the attendant to send calls to busy single-line telephones:

- 1 Type **change station *n***, where *n* is the single-line extension that you want to change. Press **Enter**.
The system displays the **Station** screen.
- 2 Press **Next** until you see the Att. Call Waiting Indication field ([Figure 17, Station screen](#), on page 157).

Figure 17: Station screen

change station 35001		Page 2 of 3
STATION		
FEATURE OPTIONS		
LWC Reception? audix	Coverage Msg Retrieval? y	
LWC Activation? y	Auto Answer: none	
LWC Log External Calls? n	Data Restriction? n	
CDR Privacy? n	Call Waiting Indication? n	
Redirect Notification? y	Att. Call Waiting Indication? y	
Per Button Ring Control? n	Distinctive Audible Alert? y	
Bridged Call Alerting? n	Adjunct Supervision? y	
Switchhook Flash? y		
Ignore Rotary Digits? y	Per Station CPN - Send Calling Number? _	
H.320 Conversion? n	Audible Message Waiting? y	
Service Link Mode: as-needed		
Multimedia Mode: basic	Coverage After Forwarding? _	
MWI Served User Type:	Multimedia Early Answer? n	
AUDIX Name: srl-aud	Direct IP-IP Audio Connections? n	
	IP Audio Hairpinning? n	
Emergency Location Ext: 75001		

- 3 Set the Att. Call Waiting Indication field to y.
- 4 Press **Enter** to save your changes.
- 5 Repeat this process for each single-line telephone on your system.

Changing the call-waiting signal

You can change the number of alerting tones (1, 2, or 3 beeps) that the user hears when the attendant extends a call. You can set the number of tones to indicate an internal call, an external call, and a priority call. The system defaults are:

- Internal: 1
- External: 2
- Priority: 3

To change the call-waiting signal:

- 1 Type **change system-parameters features**. Press **Enter**.
The system displays the *Feature-Related System Parameters* screen.
- 2 Press **Next** until you see the Distinctive Audible Alerting area ([Figure 18, Feature-Related System Parameters screen](#), on page 158).

Figure 18: Feature-Related System Parameters screen

```
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? _
Use Trunk COR for Outgoing Trunk Disconnect? _
7405ND Numeric Terminal Display? n                             7434ND? n
DISTINCTIVE AUDIBLE ALERTING
Internal: 1 External: 2 Priority: 3
Attendant Originated Calls: external
DTMF Tone Feedback Signal to VRU - Connection: _ Disconnection: _
```

- 3 Change the number of tones (1, 2, or 3 beeps) that a user hears when the system activates the Attendant Call Waiting feature.
If you change one field, **Internal**, **External**, or **Priority**, Avaya recommends that you change all fields. That way, a user knows what kind of call the attendant is extending.
- 4 Press **Enter** to save your changes.

Modifying timed intervals

If either the Timed Reminder on Hold interval, or the Return Call Timeout interval, expires before the call is answered, the call returns to the attendant console.

To modify the timed intervals:

- 1 Type **change console-parameters**. Press **Enter**.
The system displays the *Console Parameters* screen.
- 2 Press **Next** until you see the Timed Reminder on Hold field and the Return Call Timeout field ([Figure 19, Console Parameters screen](#), on page 159).

Figure 19: Console Parameters screen

change console-parameters	Page 2 of 4
CONSOLE PARAMETERS	
TIMING	
Time Reminder on Hold (sec): 30	Return Call Timeout (sec): 60
Time in Queue Warning (sec): 30	
INCOMING CALL REMINDERS	
No Answer Timeout (sec): 20	Alerting (sec): 40
Secondary Alert on Held Reminder Calls? y	
ABBREVIATED DIALING	
List1: group 1	List2:
SAC Notification? n	List3:
COMMON SHARED EXTENSIONS	
Starting Extension:	Count:

- 3 In the **Time Reminder on Hold** field, type the maximum number of seconds that a call can remain on hold before the system alerts the attendant.

In this example, we set the interval to 30 seconds.

- 4 In the **Return Call Timeout** field, type the maximum number of seconds that a call can remain on hold before the call returns to the attendant.

In this example, we set the interval to 60 seconds.

NOTE:

You must allow at least 5 seconds for each ring at all points in a coverage path. This amount of time ensures that the entire calling path is completed before the system returns the call to the console.

- 5 Press **Enter** to save your changes.

Reports for Attendant Call Waiting

The following reports provide information about the Attendant Call Waiting feature:

- None

Considerations for Attendant Call Waiting

This section provides information about how the Attendant Call Waiting feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Call Waiting under all conditions. The following considerations apply to Attendant Call Waiting:

- Attendant Call Waiting applies only to calls that are made to single-line telephones within the system.
- Only one call can wait at a time.

Interactions for Attendant Call Waiting

This section provides information about how the Attendant Call Waiting feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Call Waiting in any feature configuration.

- **Automatic Callback**
Activating Automatic Callback at the called telephone denies Attendant Call Waiting.
- **Call Coverage**
The system redirects Attendant Call Waiting calls to coverage if the called telephone has Data Privacy or Data Restriction activated. The system redirects the call to coverage if all three of the following conditions are met:
 - The Data Privacy or Data Restriction feature is activated
 - You assign call coverage to a telephone
 - The user activates Send All Calls or coverage criteria is met
 - The Coverage Don't Answer interval specifies how long that call remains directed to the called telephone before the system redirects the call to coverage. If Attendant Call Waiting is applicable on the call, the feature is active for the duration of the Don't Answer interval only. At the end of this interval, the system redirects the call 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 the attendant console. If the Don't Answer interval expires first, the system redirects the call to coverage. The system can still return the call to the attendant console if a coverage point does not answer the call before the Return Call Timeout expires.
 - If the **Station Hunting** field is assigned, and the called telephone is busy, the system redirects the call to the **Hunt To Station Assignment** value.
 - If Send All Calls is active, or if the redirection criterion is Cover All Calls, the system immediately redirects the call to coverage instead of to Attendant Call Waiting.
 - An attendant can release an extended call at any point during the call with no affect on the preceding operations.
- **Data Privacy and Data Restriction**
Activating Data Privacy or Data Restriction at the called telephone denies Attendant Call Waiting.
- **Direct Department Calling (DDC) and Uniform Call Distribution (UCD)**
Calls to a DDC group or to a UCD group do not wait. However, calls can enter the group queue, if a queue is provided.
- **Loudspeaker Paging**
Activating Loudspeaker Paging 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 that is 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 Calling of Inward Restricted Stations

Use the Attendant Calling of Inward Restricted Stations feature to allow an attendant to override an inward restriction Class of Restriction (COR).

Detailed description of Attendant Calling of Inward Restricted Stations

This section provides a detailed description of the Attendant Calling of Inward Restricted Stations feature.

A telephone that is restricted from receiving inbound calls cannot receive public network, attendant-originated, or attendant-extended calls. The Attendant Calling of Inward Restricted Stations feature allows the attendant to override this restriction.

Hardware requirements for Attendant Calling of Inward Restricted Stations

The Attendant Calling of Inward Restricted Stations feature requires the following hardware:

- An attendant console

Administering Attendant Calling of Inward Restricted Stations

The following steps are part of the administration process for the Attendant Calling of Inward Restricted Stations feature:

- [Setting up Class of Restriction override](#)

This section describes:

- Any prerequisites for administering the Attendant Calling of Inward Restricted Stations feature
- The screens that you use to administer the Attendant Calling of Inward Restricted Stations feature
- Complete administration procedures for the Attendant Calling of Inward Restricted Stations feature

Prerequisites for administering Attendant Calling of Inward Restricted Stations

You must complete the following actions before you can administer the Attendant Calling of Inward Restricted Stations feature:

- You need to set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Calling of Inward Restricted Stations

Screen name	Purpose	Fields
<i>Class of Restriction</i>	Allow an attendant to override any Class of Restriction that is assigned to a telephone.	Restriction Override

Setting up Class of Restriction override

To set up Class of Restriction override:

- 1 In the *Class of Restriction* screen, set the Restriction Override field to **attendant**. For more information, see the “Class of Restriction” feature.

Reports for Attendant Calling of Inward Restricted Stations

The following reports provide information about the Attendant Calling of Inward Restricted Stations feature:

- None

Considerations for Attendant Calling of Inward Restricted Stations

This section provides information about how the Attendant Calling of Inward Restricted Stations feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Calling of Inward Restricted Stations under all conditions. The following considerations apply to Attendant Calling of Inward Restricted Stations:

- None

Interactions for Attendant Calling of Inward Restricted Stations

This section provides information about how the Attendant Calling of Inward Restricted Stations feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Calling of Inward Restricted Stations in any feature configuration.

- None

Attendant Conference

Use the Attendant Conference feature to allow an attendant to set up a conference call for as many as six conferees, including the attendant.

Detailed description of Attendant Conference

This section provides a detailed description of the Attendant Conference feature.

The Attendant Conference feature allows an attendant to set up a conference call for as many as six conferees, including the attendant. The conference call is held at the attendant console.

To set up a conference, the attendant dials the number of a conferee, and then presses the Split button to add the party. The attendant can add parties to the conference call from both inside and outside the system.

Hardware requirements for Attendant Conference

The Attendant Conference feature requires the following hardware:

- An attendant console

Administering Attendant Conference

The following steps are part of the administration process for the Attendant Conference feature:

- [Setting up Attendant Conference](#)

This section describes:

- Any prerequisites for administering the Attendant Conference feature
- The screens that you use to administer the Attendant Conference feature
- Complete administration procedures for the Attendant Conference feature

Prerequisites for administering Attendant Conference

You must complete the following actions before you can administer the Attendant Conference feature:

- The system activates the Attendant Conference feature when you set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Conference

Screen name	Purpose	Fields
<i>Feature-Related System-Parameters</i>	Set up the Attendant Conference feature.	- Public Network Trunks on Conference Call - Conference Parties with Public Network Trunks - Conference Parties without Public Network Trunks

Setting up Attendant Conference

To set up Attendant Conference:

- 1 Type **change system-parameters features**. Press **Enter**.
The system displays the *Feature-Related System-Parameters* screen.
- 2 Press **Next** until you see the Public Network Trunks on Conference Call field ([Figure 20, Feature-Related System-Parameters screen](#), on page 166).

Figure 20: Feature-Related System-Parameters screen

change system-parameters features	Page 6 of 14
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): 0	Mode Code Interface? y
Reset Shift Timer (seconds): 0	
Station Call Transfer Recall Timer (seconds): 0	
DID Busy Treatment: tone	
Invalid Number Dialed Intercept Treatment: tone	
Allow AAR/ARS Access from DID/DIOD? y	
Allow ANI Restriction on AAR/ARS? n	
Use Trunk COR for Outgoing Trunk Disconnect? n	
7405ND Numeric Terminal Display? y	7434ND? n
DISTINCTIVE AUDIBLE ALERTING	
Internal: 1 External: 2 Priority: 3	
Attendant originated Calls: external	
DTMF Tone Feedback Signal to VRU - Connection:	Disconnection:

- 3 In the **Public Network Trunks on Conference Call** field, type a number between 0 and 5. This number indicates the maximum number of public network trunks that are allowed on an attendant conference call.

In this example, we set the number to 5. If you set this field to 0, the **Conference Parties with Public Network Trunks** field does not appear.
- 4 In the **Conference Parties with Public Network Trunks** field, type a number between 3 and 6. This number indicates the maximum number of parties allowed on a conference call that involves a public network subscriber.

In this example, we set the number to 6.
- 5 In the **Conference Parties without Public Network Trunks** field, type a number between 3 and 6. This number indicates the maximum number of parties allowed on a conference call that involves no public network trunks.

In this example, we set the number to 6.
- 6 Press **Enter** to save your changes.

Reports for Attendant Conference

The following reports provide information about the Attendant Conference feature:

- None

Considerations for Attendant Conference

This section provides information about how the Attendant Conference feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Conference under all conditions. The following considerations apply to Attendant Conference:

- Use this feature whenever an attendant needs to talk with more than one party at the same time, or to establish a conference call for parties who are outside the system.
- The attendant can set up only one conference call at a time. The attendant can hold a conference call on the console, or release the conference call.
- The attendant cannot handle other calls while setting up a conference call.
- No matter who established the call, once the attendant is one of the conferees, only the attendant can add another party to the call.

Interactions for Attendant Conference

This section provides information about how the Attendant Conference feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Conference in any feature configuration.

- Bridged Call Appearance

A user can press the Bridged Appearance button on the telephone and join a conference call if no more than 5 parties are on the call.

- Call Vectoring

A VDN can be included as a party in a conference call only after vector processing terminates for that call. Vector processing terminates, for example, after a successful route-to command.

- Trunk-to-Trunk Transfer

If Trunk-to-Trunk Transfer is disabled on the *Feature-Related System-Parameters* screen, and the attendant releases from a conference call that involves only trunk conferees, the system disconnects the trunks.

When a user of a multifunction telephone (BRI/Digital/Hybrid) dials enough digits to route a call that might otherwise be routed differently if the user dials additional digits, the telephone does not recognize the Conference or Transfer buttons. For the telephone to recognize the Conference or Transfer buttons, the user must either delays dialing for 3-seconds, or dial the pound key (#). This action tells the system to route the call based on the digits that were already dialed, and not wait for additional digits. The system completes the call.

Attendant conferencing might not operate properly if the central office (CO) does not provide answer supervision. In this case, the Answer Supervision Timeout and Outgoing End of Dial fields are set to the same nonzero number, and the Receive Answer Supervision field is set to **n**.

If the CO provides answer supervision, you can set the Answer Supervision Timeout field to **0**, and the Receive Answer Supervision field to **y**.

Attendant Control of Trunk Group Access

Use the Attendant Control of Trunk Group Access feature to allow the attendant to control outgoing and two-way trunk groups. The attendant usually activates this feature during periods of high use.

This feature also prevents telephone users from directly accessing an outgoing trunk group that the attendant has controlled.

Detailed description of Attendant Control of Trunk Group Access

This section provides a detailed description of the Attendant Control of Trunk Group Access feature.

When an administered threshold for a trunk group is reached, a warning lamp for that trunk group lights on the attendant console. The attendant can then access control of that trunk group. To gain direct access to an outgoing trunk group, the attendant presses the button that is assigned to that trunk group.

When the Attendant Control of Trunk Group Access feature is activated:

- Internal callers who use a trunk access code to dial out are connected to the attendant
- The attendant can prioritize outgoing calls for the remaining trunks

The system processes Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) route pattern calls without the need for attendant control.

Each attendant console has 12 designated Trunk Hundreds Select buttons. These buttons can be administered for the Attendant Control of Trunk Group Access feature. You can also administer each console with up to 12 feature buttons for Trunk Hundreds Select buttons, which gives you up to 24 buttons to use for this feature.

Each Trunk Hundreds Select button has a busy lamp. The busy lamp lights when the telephones of all members of the trunk group are busy.

NOTE:

If you administer one of the two-lamp feature buttons on a basic console as a Trunk Hundreds Select button, use the bottom lamp as the busy lamp.

The Trunk Hundreds Select buttons have two additional lamps for Attendant Control of Trunk Group Access. The two lamps are:

- Warn (warning) lamp

The 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

The control lamp lights when the attendant activates the Attendant Control of Trunk Group Access feature for the associated trunk group. You assign the **act-tr-grp** and **deact-tr-g** buttons on the **Attendant Console** screen to allow the attendant to activate and deactivate the trunk group access.

Hardware requirements for Attendant Control of Trunk Group Access

The Attendant Control of Trunk Group Access feature requires the following hardware:

- An attendant console

Administering Attendant Control of Trunk Group Access

The following steps are part of the administration process for the Attendant Control of Trunk Group Access feature:

- [Setting the trunk group threshold](#)
- [Assigning Attendant Control of Trunk Group Access buttons](#)

This section describes:

- Any prerequisites for administering the Attendant Control of Trunk Group Access feature
- The screens that you use to administer the Attendant Control of Trunk Group Access feature
- Complete administration procedures for the Attendant Control of Trunk Group Access feature

Prerequisites for administering Attendant Control of Trunk Group Access

You must complete the following actions before you can administer the Attendant Control of Trunk Group Access feature:

- Set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Control of Trunk Group Access

Screen name	Purpose	Fields
<i>Trunk Group</i>	Determine the threshold for when to light the warning lamp.	Busy Threshold
<i>Attendant Console</i>	Assign buttons to allow the attendant to activate and deactivate trunk group access.	Any unassigned buttons in the Feature Button Assignments area.

Setting the trunk group threshold

To set the trunk group threshold:

- 1 Type **change trunk-group *n***, where *n* is the number of the trunk group. Press **Enter**.
The system displays the *Trunk Group* screen ([Figure 21, Trunk Group screen](#), on page 171).

Figure 21: Trunk Group screen

change trunk-group 4
Page 1 of x

TRUNK GROUP

Group Number: 4

Group Name: _____

Direction: _____

Dial Access? _

Queue Length: ____

Comm Type: _____

BCC: _

Group Type: _____

COR: _

Outgoing Display? _

Busy Threshold: ____

Auth Code? _

Trunk Flash? _

ITC? _____

CDR Reports: _

TN: ____

TAC: ____

Trunk Signaling Type: ____

Night Service: _____

Incoming Destination: _____

TRUNK PARAMETERS

Trunk Type (in/out): _____

Outgoing Dial Type: _____

Digit Treatment: _____

Analog Loss Group: ____

Incoming Dial Tone? _

Incoming Calling Number - Delete: ____

Bit Rate: _____

Disconnect Supervision - In? _

Answer Supervision Timeout: ____

Incoming Rotary Timeout(sec): ____

Incoming Dial Type: _____

Disconnect Timing(msec): _____

Digits: _____

Sig Bit Inversion: none

Digital Loss Group: ____

Insert: _____

Synchronization: _____

Out? _

Receive Answer Supervision? _

Format: _____

Duplex: _____

- 2 In the **Busy Threshold** field, type the number of trunks that must be busy to light the warning lamp on the attendant console. The range is from 1 to 255.

For example, a trunk group contains 30 trunks. If you want to alert the attendant when 25 or more trunks are in use, type **25**.
- 3 Press **Enter** to save your changes.

Assigning Attendant Control of Trunk Group Access buttons

To assign Attendant Control of Trunk Group Access buttons:

- 1 Type **change attendant *n***, where *n* is the number of the attendant console. Press **Enter**.
The system displays the **Attendant Console** screen.
- 2 Press **Next** until you see the Feature Button Assignments area ([Figure 22, Attendant Console screen](#), on page 172).

Figure 22: Attendant Console screen

change attendant 1		ATTENDANT CONSOLE		Page 3 of 4
FEATURE BUTTON ASSIGNMENTS				
1: split_____		13: _____		
2: _____		14: _____		
3: act-tr-grp		15: _____		
4: deact-tr-g		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_ *		

- 3 In the Feature Button Assignments area, assign **act-tr-grp** and **deact-tr-g** to two available buttons.
In this example, we assign **act-tr-grp** to button **3**, and **deact-tr-g** to button **4**.
- 4 Press **Enter** to save your changes.

For more information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Reports for Attendant Control of Trunk Group Access

The following reports provide information about the Attendant Control of Trunk Group Access feature:

- None

Considerations for Attendant Control of Trunk Group Access

This section provides information about how the Attendant Control of Trunk Group Access feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Control of Trunk Group Access under all conditions. The following considerations apply to Attendant Control of Trunk Group Access:

- None

Interactions for Attendant Control of Trunk Group Access

This section provides information about how the Attendant Control of Trunk Group Access feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Control of Trunk Group Access in any feature configuration.

- **Authorization Codes**
Authorization codes do not collect when a trunk group has an incoming destination set to the attendant.
- **Automatic Route Selection (ARS) and Automatic Alternate Routing (AAR)**
Activating Attendant Control of Trunk Group Access removes the controlled trunk groups from the ARS and the AAR patterns. Deactivating the feature reinserts the groups into the patterns. The system does not route ARS calls to the attendant.
- **QSIG**
QSIG trunks do not support Attendant Control of Trunk Group Access.
- **Uniform Dial Plan (UDP)**
Activating Attendant Control of Trunk Group Access removes the controlled trunk groups from preferences. Deactivating the feature allows the UDP to access the trunk groups.

Attendant Direct Extension Selection

Use the Attendant Direct Extension Selection (DXS) feature to allow an attendant to track the idle or busy status of an extension. This feature also allows an attendant to place or extend calls to extensions without the need to dial the extension.

The Attendant DXS feature is sometimes referred to as the Attendant Direct Extension Selection with Busy Lamp Field feature.

Attendant DXS supports the following capabilities:

- Standard DXS Tracking
- Enhanced DXS Tracking
- Group Display button

Detailed description of Attendant Direct Extension Selection

This section provides a detailed description of the Attendant Direct Extension Selection (DXS) feature.

With Attendant DXS, you can use either Standard or Enhanced DXS Tracking:

- Standard DXS Tracking

If your attendant console has one or more Hundreds Select buttons, the attendant can press both a Hundreds Select button and a DXS button to access an extension.

- Enhanced DXS Tracking

Use Enhanced DXS Tracking if your attendant console:

- Does not use Hundreds Select buttons
- Uses Hundreds Select buttons, but you have one or more hundreds groups not administered by a Hundreds Select button

To access an extension, the attendant can:

- Press a Group Select button
- Dial the first 2 or 3 digits of the extension
- Press the DXS button to access an extension

When the system tracks a group of extensions, the attendant can press the DXS button to place or extend subsequent calls to extensions in that group without having to reselect the group. Whether the attendant uses a Hundreds Select button or the Group Select button, both capabilities eliminate the need to dial extensions.

Extensions might be telephones, hunt-group extensions, off-switch extensions such as uniform dial plan (UDP) extensions, or other extensions.

Whichever method you use to access and track DXS extensions, you can use a Group Display feature button to view the group of extensions currently being tracked. This button on the console indicates the range of extensions that the selector console is tracking.

NOTE:

An associated DXS lamp for a vector directory number (VDN) is always dark. The attendant can use the DXS button to place a call to a VDN.

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, the total number of Hundreds Select buttons for each attendant, including both attendant console feature buttons and selector console buttons, cannot exceed 20.

Enhanced DXS Tracking

Enhanced DXS Tracking is helpful if you have more than 100 telephones, but use a console that does not have Hundreds Select buttons administered. Enhanced DXS Tracking is also helpful if you have more telephones than 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. The attendant uses this button 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 3 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 that are 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 presses this button, the system identifies the digits that are associated with a Hundreds Select button. If no Hundreds Select button is lit, the system identifies the digits that were 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.

Hardware requirements for Attendant Direct Extension Selection

The Attendant Direct Extension Selection (DXS) feature requires the following hardware:

- An attendant console

Administering Attendant Direct Extension Selection

This section describes the prerequisites and the screens for the Attendant Direct Extension Selection (DXS) feature.

Prerequisites for administering Attendant Direct Extension Selection

You must complete the following actions before you can administer the Attendant Direct Extension Selection (DXS) feature:

- Set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Direct Extension Selection

Screen name	Purpose	Fields
<i>Attendant Console</i>	Assign a DXS button.	Any available button field in the HUNDREDS SELECT BUTTON ASSIGNMENTS area

For more information, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Reports for Attendant Direct Extension Selection

The following reports provide information about the Attendant Direct Extension Selection feature:

- None

Considerations for Attendant Direct Extension Selection

This section provides information about how the Attendant Direct Extension Selection (DXS) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Direct Extension Selection (DXS) under all conditions. The following considerations apply to Attendant Direct Extension Selection (DXS):

- With this feature, the attendant can place calls to:
 - 800 extensions with the basic selector console
 - 2,000 extensions with the enhanced selector console
 - up to 99,999 extensions with the Group Select feature button (extension numbers from 100 to 99,999)

If the attendant is tracking a hundreds group with either a Hundreds Select button or the Group Select feature button, the attendant presses the DXS button to access an extension.

- This feature provides the attendant with a visual indication of the idle or busy status of the extensions assigned to the selected hundreds group. You can monitor up to 100 extensions for idle or busy status at a time.
- Enhanced DXS Tracking does not support extensions with fewer than 3 digits.
- Extension tracking is possible only for the system on which the attendant resides.

Interactions for Attendant Direct Extension Selection

This section provides information about how the Attendant Direct Extension Selection (DXS) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Direct Extension Selection (DXS) in any feature configuration.

- **Attendant Display**
When the attendant uses the Attendant Direct Extension Selection with Busy Lamp Field, the alphanumeric display identifies the call through the Attendant Display.
- **Call Coverage**
If Send All Calls is activated, or if the Call Coverage redirection criteria are met, then the system redirects an extended call to the coverage path.
- **Centralized Attendant Service (CAS)**
When the attendant uses a DXS button to make a CAS call, the attendant hears ringback tone after a few seconds elapse.

Attendant Direct Trunk Group Selection

Use the Attendant Direct Trunk Group Selection feature to allow the attendant direct access to an idle outgoing trunk.

Detailed description of Attendant Direct Trunk Group Selection

This section provides a detailed description of the Attendant Direct Trunk Group Selection feature.

With this feature, the attendant gets direct access to an idle outgoing trunk when the attendant presses the button that is assigned to the trunk group. This feature eliminates the need for the attendant to memorize, or look up, and dial the trunk access codes (TACs) associated with frequently used trunk groups.

You can use up to 12 designated Trunk Hundreds Select buttons on each console. You can also administer up to 12 of the feature buttons as additional Trunk Hundreds Select buttons, for a total of 24 Trunk Hundreds Select buttons per console. Each button allows the attendant direct access to an outgoing select trunk group.

While an attendant talks on a call, the attendant can be split away that call and place a new call to the outgoing trunk that is specified by the trunk group select button. The attendant can then press Release to connect the split-away parties to the dial tone on the trunk. Or the attendant can dial the destination and press Release to connect the split-away party to the called party.

All Trunk Hundreds Select buttons, including any administered on the feature buttons, have a Busy lamp. This lamp lights up when all trunks in the associated trunk group are busy. If you administer one of the two-lamp feature buttons on a basic console as a Trunk Hundreds Select button, use the bottom lamp as the Busy lamp. Six of the designated buttons on a basic console, or all 12 designated buttons on an enhanced console also have a Warning lamp and a Control lamp. The Warning lamp lights when a preset number of trunks in the associated trunk group are busy. The Cont lamp lights when the attendant activates Attendant Control of Trunk Group Access for the associated trunk group.

You can assign Loudspeaker Paging zones instead of trunk groups to Trunk Hundreds Select buttons. The Busy lamp then indicates the idle or the busy status of the associated Loudspeaker Paging zone.

Hardware requirements for Attendant Direct Trunk Group Selection

The Attendant Direct Trunk Group Selection feature requires the following hardware:

- An attendant console

Administering Attendant Direct Trunk Group Selection

This section describes the prerequisites and the screens for the Attendant Direct Trunk Group Selection feature.

Prerequisites for administering Attendant Direct Trunk Group Selection

You must complete the following actions before you can administer the Attendant Direct Trunk Group Selection feature:

- Set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Direct Extension Selection

Screen name	Purpose	Fields
<i>Attendant Console</i>	Assign trunk access codes (TACs) for local and remote systems.	Any available button field in the DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes) area

For more information, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Reports for Attendant Direct Trunk Group Selection

The following reports provide information about the Attendant Direct Trunk Group Selection feature:

- None

Considerations for Attendant Direct Trunk Group Selection

This section provides information about how the Attendant Direct Trunk Group Selection feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Direct Trunk Group Selection under all conditions. The following considerations apply to Attendant Direct Trunk Group Selection:

- Attendant Direct Trunk Group Selection eliminates the need for the attendant to memorize, or look up, and dial a trunk access code (TAC) that is associated with frequently used trunk groups. A label that is associated with each Trunk Hundreds Select button identifies the destination or use of the button. For example, buttons might be labeled Chicago, FX, or WATS. Pressing the button selects an idle trunk in the desired group.

Interactions for Attendant Direct Trunk Group Selection

This section provides information about how the Attendant Direct Trunk Group Selection feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Direct Trunk Group Selection in any feature configuration.

- Attendant Control of Trunk Group Access
If Attendant Control of Trunk Group Access is provided, the Attendant Direct Trunk Group Selection feature must also be provided.
- QSIG
Attendant Direct Trunk Group Selection does not apply to QSIG trunks.

Attendant Intrusion

Use the Attendant Intrusion feature to allow an attendant to intrude on an existing call. The Attendant Intrusion feature is also called Call Offer.

Detailed description of Attendant Intrusion

This section provides a detailed description of the Attendant Intrusion feature.

The attendant uses the Attendant Intrusion feature to announce a new call or a message to a user who is already on a call.

When the attendant releases the call of the user, the source party either waits at the analog telephone of the user, or holds on an available line appearance on a digital telephone.

NOTE:

Only one call can be waiting at a time. If a call is already waiting on the telephone of the intruded party, the source party, once split from the attendant, cannot also wait.

Hardware requirements for Attendant Intrusion

The Attendant Intrusion feature requires the following hardware:

- An attendant console

Administering Attendant Intrusion

The following steps are part of the administration process for the Attendant Intrusion feature:

- [Assigning an intrusion button](#)

This section describes:

- Any prerequisites for administering the Attendant Intrusion feature
- The screens that you use to administer the Attendant Intrusion feature
- Complete administration procedures for the Attendant Intrusion feature

Prerequisites for administering Attendant Intrusion

You must complete the following actions before you can administer the Attendant Intrusion feature:

- Set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Intrusion

Screen name	Purpose	Fields
<i>Attendant Console</i>	Assign an intrusion button.	Any unassigned button in the Feature Button Assignments area.

Assigning an intrusion button

To assign an **intrusion** button:

- 1 Type **change attendant *n***, where *n* is the number of the attendant console. Press **Enter**.
The system displays the *Attendant Console* screen.
- 2 Press **Next** until you see the Feature Button Assignments area ([Figure 23, Attendant Console screen](#), on page 184).

Figure 23: Attendant Console screen

change attendant 1		ATTENDANT CONSOLE		Page 3 of 4
FEATURE BUTTON ASSIGNMENTS				
1: split		13: _____		
2: dont-split		14: _____		
3: act-tr-grp		15: _____		
4: deact-tr-g		16: _____		
5: intrusion		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_ *		

- 3 In the Feature Button Assignments area, assign **intrusion** to an available button.
In this example, we assign **intrusion** to button 5.
- 4 Press **Enter** to save your changes.

For more information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Reports for Attendant Intrusion

The following reports provide information about the Attendant Intrusion feature:

- None

Considerations for Attendant Intrusion

This section provides information about how the Attendant Intrusion feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Intrusion under all conditions. The following considerations apply to Attendant Intrusion:

- None

Interactions for Attendant Intrusion

This section provides information about how the Attendant Intrusion feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Intrusion in any feature configuration.

- Intrusion is denied in when:
 - A telephone is on a conference call with the administered maximum number of conferees
 - A call is established with Data Privacy activated
 - A call is established 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 waiting for an intruded party, the second caller, who is split from the attendant, cannot use Call Waiting to wait for the intruded party. The attendant display shows 'wait' or 'busy'. If an intrusion is possible, the attendant display shows '1 wait' or '1 busy'.
- In Italy only, the system provides Attendant Intrusion on remote telephones through TGU/TGE trunks.

Attendant Intrusion

Interactions for Attendant Intrusion

Attendant Lockout – Privacy

Use the Attendant Lockout – Privacy feature to prevent an attendant, who drops from a multiparty conference call, from reentering the call.

Detailed description of Attendant Lockout – Privacy

This section provides a detailed description of the Attendant Lockout – Privacy feature.

The Attendant Lockout – Privacy feature provides privacy for parties on a multiparty call. The parties can hold a private conversation without interruption by the attendant. The multiparty call must be held on the attendant console. The attendant can be recalled only by a telephone user who is on the multiparty call.

You administer this features on a system-wide basis. The Attendant Lockout – Privacy feature is either activated or deactivated.

Hardware requirements for Attendant Lockout – Privacy

The Attendant Lockout – Privacy feature requires the following hardware:

- An attendant console

Administering Attendant Lockout – Privacy

The following steps are part of the administration process for the Attendant Lockout – Privacy feature:

- [Activating or deactivating the Attendant Lockout – Privacy feature](#)

This section describes:

- Any prerequisites for administering the Attendant Lockout – Privacy feature
- The screens that you use to administer the Attendant Lockout – Privacy feature
- Complete administration procedures for the Attendant Lockout – Privacy feature

Prerequisites for administering Attendant Lockout – Privacy

You must complete the following actions before you can administer the Attendant Lockout – Privacy feature:

- Set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Lockout – Privacy

Screen name	Purpose	Fields
<i>Console Parameters</i>	Activate or deactivate the Attendant Lockout – Privacy feature.	Attendant Lockout

Activating or deactivating the Attendant Lockout – Privacy feature

To activate or deactivate the Attendant Lockout – Privacy feature:

- 1 Type **change console-parameters**. Press **Enter**.

The system displays the *Console Parameters* screen ([Figure 24, Console Parameters screen](#), on page 189).

Figure 24: Console Parameters screen

change console-parameters

Page 1 of 3

CONSOLE PARAMETERS

Attendant Group Name: 27 character name OPERATOR

COS: 0

COR: 0

Calls in Queue Warning: 5

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):

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 Attendant Lockout field, perform one of the following actions:

- Type **y** to activate this feature.
- Type **n** to deactivate this feature.

3 Press **Enter** to save your changes.

For more information, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Reports for Attendant Lockout – Privacy

The following reports provide information about the Attendant Lockout – Privacy feature:

- None

Considerations for Attendant Lockout – Privacy

This section provides information about how the Attendant Lockout – Privacy feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Lockout – Privacy under all conditions. The following considerations apply to Attendant Lockout – Privacy:

- None

Interactions for Attendant Lockout – Privacy

This section provides information about how the Attendant Lockout – Privacy feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Lockout – Privacy in any feature configuration.

- **Attendant Recall**

If Attendant Lockout – Privacy is activated, use Attendant Recall if you must recall the attendant.

- **Individual Attendant Access**

Attendant Lockout – Privacy applies only to attendant group calls. This feature does not affect individual attendant calls.

- **Trunk-to-Trunk Transfer**

Attendant Lockout – Privacy does not function when a call that uses Trunk-to-Trunk Transfer is held on the console.

Attendant Override of Diversion Features

Use the Attendant Override of Diversion Features feature to allow the attendant to bypass a call-diverting feature that a user activates. Such features include Send All Calls, Call Coverage, and Call Forwarding.

Detailed description of Attendant Override of Diversion Features

This section provides a detailed description of the Attendant Override of Diversion Features feature.

With the Attendant Override of Diversion Features, the attendant can bypass a call-diverting feature that a user activates. Such features include Send All Calls, Call Coverage, and Call Forwarding.

An attendant can use this feature, in combination with Attendant Intrusion, to place an emergency call or an urgent call to a user.

Hardware requirements for Attendant Override of Diversion Features

The Attendant Override of Diversion Features feature requires the following hardware:

- An attendant console

Administering Attendant Override of Diversion Features

This section contains prerequisites and the screens for administering the Attendant Override of Diversion Features feature.

Prerequisites for administering Attendant Override of Diversion Features

You must complete the following actions before you can administer the Attendant Override of Diversion Features feature:

- Set up an attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Override of Diversion Features

Screen name	Purpose	Fields
<i>Attendant Console</i>	Administer the Attendant Override button.	Any available button field in the FEATURE BUTTON ASSIGNMENTS area

Reports for Attendant Override of Diversion Features

The following reports provide information about the Attendant Override of Diversion Features feature:

- None

Considerations for Attendant Override of Diversion Features

This section provides information about how the Attendant Override of Diversion Features feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Override of Diversion Features under all conditions. The following considerations apply to Attendant Override of Diversion Features:

- None

Interactions for Attendant Override of Diversion Features

This section provides information about how the Attendant Override of Diversion Features feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Override of Diversion Features in any feature configuration.

- None

Attendant Priority Queue

Use the Attendant Priority Queue feature to place incoming calls to an attendant, that cannot be immediately answered, into an ordered queue.

Attendant Priority Queue supports the following capabilities:

- Priority by Call Category
- Priority by Call Type

Detailed description of Attendant Priority Queue

This section provides a detailed description of the Attendant Priority Queue feature.

When an attendant cannot immediately answer calls, the system can place incoming calls into categories, or into a call type within a category. With this feature, you can define 13 different categories of incoming attendant calls. Emergency calls always have the highest priority.

With Attendant Priority Queue, attendants can answer calls by call category, such as trunk type. Attendant Priority Queue places incoming calls in a queue:

- according to the priority levels that you assign for each type of call
- in order of time stamp within each level

The calling party hears ringback until an attendant answers the call.

Priority by Call Category

Assign an Attendant Priority Queue level to each of 13 incoming attendant call categories. Each category has a default level. You can reset the priority level for any category.

Important:

Do not change the priority level for Emergency Access calls. Emergency Access calls must remain priority 1.

The attendant call categories are:

- Emergency Access. Calls from users who dial the emergency access code. The default is the highest priority level. This category must remain priority 1.
- Assistance Call. Calls from users who:
 - dial the attendant group access code
 - have the Manual Originating Line Service feature activated
- CO Call. Incoming central office (CO), foreign exchange (FX), or Wide Area Telecommunications Service (WATS) trunk calls to an attendant group. This category does not including trunk calls that are returned to the attendant group after a timeout or a deferred attendant recall.

Attendant Priority Queue

Detailed description of Attendant Priority Queue

- DID to Attendant. Incoming direct inward dial (DID) trunk calls to an attendant group.
- Tie Call. Incoming Tie trunk calls, including dial-repeating and direct types.
- Redirected DID Call. DID or Automatic Call Distribution (ACD) calls that time out, and are rerouted to the attendant group. The timeout can be caused by:
 - ring/no-answer
 - busy condition
 - Number Unobtainable
- Redirected Call. Calls that are assigned to one attendant, but redirected to the attendant group because the attendant is now busy.
- Return Call. Calls that are returned to the attendant after the calls time out. If the attendant is now busy, calls are redirected to the attendant group.
- Serial Call. Calls from the Attendant Serial Call feature. Calls fit this category when an outside trunk call, that the attendant designates as a serial call, is extended to and completed at a telephone, and then the user goes on-hook. If the attendant who extended the call is busy, the call redirects to the attendant group.
- Individual Attendant Access. Calls from users, incoming trunks, or a system feature, to the Individual Attendant Access (IAA) extension of a specific attendant. If the attendant is busy, the call remains in a queue until the attendant is available.
- Interpositional. Calls from one attendant to the Individual Attendant Access (IAA) extension of another attendant.
- VIP Wakeup Reminder Call. Call from the Hospitality feature that send a wake-up reminder to the attendant to call the room.
- Miscellaneous Calls. All other calls.

You can assign the same priority level to more than one category. Assigning all categories the same priority level creates a first-in first-out queue.

When at least one call appears in the Attendant Priority Queue, the Calls Waiting lamp lights steadily on all active attendant consoles. If the number of calls in the queue reaches the attendant-group calls-waiting threshold, the Queue Warning lamp lights steadily on all active attendant consoles.

Priority by Call Type

You can further define the priority that you assign to calls in the Attendant Priority Queue by call type. Then, within each call type, prioritize calls by time.

The call types, in descending order of priority, are:

- Type 1 calls are outgoing public-network calls that receive answer supervision when the Answer Supervision Timer of the trunk group expires, even if the trunk is still ringing. Type 1 calls are also incoming calls that the attendant answers.
- Type 2 calls are incoming external public-network calls before the calls receive answer supervision, or before the Answer Supervision Timer of the trunk group expires.

- Type 3 calls are all other calls, which include internal calls, conference calls, and tie-trunk calls of any type.

NOTE:

External public-network calls have priority over all other calls, including conference calls.
Answered public-network calls have priority over calls that are not yet answered.

Hardware requirements for Attendant Priority Queue

The Attendant Priority Queue feature requires the following hardware:

- An attendant console

Administering Attendant Priority Queue

The following steps are part of the administration process for the Attendant Priority Queue feature:

- [Setting category priorities](#)
- [Setting the number of calls in the queue](#)
- [Assigning a Call Type button](#)
- [Translating the Call Type button into a user-defined language](#)

This section describes:

- Any prerequisites for administering the Attendant Priority Queue feature
- The screens that you use to administer the Attendant Priority Queue feature
- Complete administration procedures for the Attendant Priority Queue feature

Prerequisites for administering Attendant Priority Queue

You must complete the following actions before you can administer the Attendant Priority Queue feature:

- Set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Priority Queue

Screen name	Purpose	Fields
<i>Console Parameters</i>	Assign priorities to queue categories.	All fields in the Queue Priorities area
<i>Feature-Related System Parameters</i>	Set the number of calls in the queue.	Reserved Slots for Attendant Priority Queue
<i>Attendant Console</i>	Assign a Call Type button.	Any unassigned button in the Feature Button Assignments area
<i>Language Translations</i>	Translate the Call Type button into a user-defined language.	Call Type

Setting category priorities

To set the category priorities:

- 1 Type **change console-parameters**. Press **Enter**.
The system displays the *Console Parameters* screen.
- 2 Press **Next** until you see the **Queue Priorities** area ([Figure 23, Console Parameters screen](#), on page 142).

Figure 25: Console Parameters screen

change console-parameters	Page 3 of 4
CONSOLE PARAMETERS	
QUEUE PRIORITIES	
Emergency Access:1_	
Assistance Call:2_	
CO Call:3_	
DID to Attendant:4_	
Tie Call:5_	
Redirected DID Call:6_	
Redirected Call:7_	
Return Call:8_	
Serial Call:9_	
Individual Attendant Access:10	
Interpositional:11	
VIP Wakeup Reminder Call:12	
Miscellaneous Call:13	
Call-Type Ordering Within Priority Levels? y	

- 3 Type a number from 1 to 13 next to each category.
This number indicates the priority of the category. The number 1 indicates the highest priority. The number 13 indicates the lowest priority.

- 4 In the Call-Type Ordering Within Priority Levels field, perform one of the following actions:
 - Type **y** if you want to order calls by call type within each category.
 - Type **n** if you do not want to order calls by call type within each category.
- 5 Press **Enter** to save your changes.

Setting the number of calls in the queue

To set the number of calls in the queue:

- 1 Type **change system-parameters features**. Press **Enter**.
The system displays the *Feature-Related System Parameters* screen
- 2 Press **Next** until you see the Reserved Slots for Attendant Priority Queue field ([Figure 24, Feature-Related System Parameters screen](#), on page 143).

Figure 26: Feature-Related System Parameters screen

change system-parameters features	Page 4 of 14
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: 5_	
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? n	
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? y
Controlled Toll Restriction Replaces: none	

- 3 In the Reserved Slots for Attendant Priority Queue field, type a number between 2 and 75. This number indicates how many non emergency calls can go into the priority queue. The default is 5.
- 4 In the Number of Emergency Calls Allowed in Attendant Queue field, type a number between 0 and 75. This number indicates how many emergency calls can go into the priority queue. The default is 5.
- 5 Press **Enter** to save your changes.

Assigning a Call Type button

You can assign a Call Type button on the attendant console. When you press this button, the system displays the call type of the active call. For more information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Translating the Call Type button into a user-defined language

If you use a user-defined language to display messages on the attendant console, you can translate the Call Type button.

To translate the Call Type button into a user-defined language:

- 1 Type **change display-messages miscellaneous-features**. Press **Enter**.
The system displays the *Language Translations* screen.
- 2 Press **Next** until you see the English: Call Type field ([Figure 25, Language Translations screen](#), on page 144).

Figure 27: Language Translations screen

change display-messages miscellaneous-features	Page 5 of 6
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: *****	

- 3 In the English: Call Type field, translate the button label into the user-defined language.
- 4 Press **Enter** to save your changes.

For more information, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Reports for Attendant Priority Queue

The following reports provide information about the Attendant Priority Queue feature:

- None

Considerations for Attendant Priority Queue

This section provides information about how the Attendant Priority Queue feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Priority Queue under all conditions. The following considerations apply to Attendant Priority Queue:

- An incoming call that defaults to an attendant and is then redirected to the attendant group, does not change the associated Attendant Priority Queue level. The reason code that displays on the answering attendant console remains the same as the reason code that displays on the original attendant console.

Interactions for Attendant Priority Queue

This section provides information about how the Attendant Priority Queue feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Priority Queue in any feature configuration.

- **Multiparty Calls**
The system always treats multiparty calls as Type 3 calls. If a multiparty call becomes a single-party call while in the queue, the multiparty call remains a Type 3 call.
- **Night Service-Hunt Group**
When you use Night Service-Hunt Groups, retrieve calls from the hunt groups instead of from the Attendant Priority Queue. Since call-type prioritization does not apply to hunt groups, do not retrieve calls in order of call type, unless you designate the Attendant Priority Queue as the termination.
- **Off-Premises Station**
The system always identifies calls from off-premises telephones as Type 3 calls.

Attendant Recall

Use the Attendant Recall feature to allow a user to recall the attendant while the user is on a call.

Detailed description of Attendant Recall

This section provides a detailed description of the Attendant Recall feature.

With the Attendant Recall feature, users can call the attendant for assistance while the users are currently on a call. Users can activate the Attendant Recall feature only when they are on a:

- Two-party call
- Conference call that is held at the attendant console

Hardware requirements for Attendant Recall

The Attendant Recall feature requires the following hardware:

- An attendant console

Administering Attendant Recall

The following steps are part of the administration process for the Attendant Recall feature:

- The system activates the Attendant Recall feature when you set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

This section describes:

- Any prerequisites for administering the Attendant Recall feature
- The screens that you use to administer the Attendant Recall feature
- Complete administration procedures for the Attendant Recall feature

Prerequisites for administering Attendant Recall

You must complete the following actions before you can administer the Attendant Recall feature:

- None

Screens for administering Attendant Recall

Screen name	Purpose	Fields
<i>Attendant Console</i>	Set up an attendant console.	All

End-user procedures for Attendant Recall

End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Attendant Recall:

To recall the attendant from your telephone:

- Single-line users – Press the recall button or flash the switchhook.
- Multiappearance users – Press the conference button or the transfer button.

Reports for Attendant Recall

The following reports provide information about the Attendant Recall feature:

- None

Considerations for Attendant Recall

This section provides information about how the Attendant Recall feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Recall under all conditions. The following considerations apply to Attendant Recall:

- None

Interactions for Attendant Recall

This section provides information about how the Attendant Recall feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Recall in any feature configuration.

- Individual Attendant Access

If a user is holding a hunt-group call to an individual attendant, the user who is active on the call cannot recall the attendant. However, the user can transfer calls and make conference calls.

Attendant Serial Calling

Use the Attendant Serial Calling feature to allow the attendant to transfer trunk calls that the system routes to the attendant when the called user hangs up. The attendant can then transfer the call to another user who is on the same server.

Detailed description of Attendant Serial Calling

This section provides a detailed description of the Attendant Serial Calling feature.

You can administer the system to route calls to the attendant when the called user hangs up, but the caller does not. The attendant can then use Attendant Serial Calling to transfer the call to another user, at the request of the caller. The attendant can transfer calls only to those users on the same server.

This feature helps the attendant to use the trunks efficiently. If few trunks are available, and direct inward dialing (DID) is unavailable, a caller might make several attempts to complete a call before the caller is successful. When a caller makes a successful call to the attendant, the user can use the same line into the server for multiple telephone calls.

The attendant display shows the attendant that an incoming call is a serial call.

Hardware requirements for Attendant Serial Calling

The Attendant Serial Calling feature requires the following hardware:

- An attendant console

Administering Attendant Serial Calling

This section contains prerequisites and the screens for administering the Attendant Serial Calling feature.

Prerequisites for administering Attendant Serial Calling

You must complete the following actions before you can administer the Attendant Serial Calling feature:

- Set up an attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Serial Calling

Screen name	Purpose	Fields
<i>Attendant Console</i>	Administer the button for the Attendant Serial Call feature. serial-cal	Any available button field in the FEATURE BUTTON ASSIGNMENTS area

Reports for Attendant Serial Calling

The following reports provide information about the Attendant Serial Calling feature:

- None

Considerations for Attendant Serial Calling

This section provides information about how the Attendant Serial Calling feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Serial Calling under all conditions. The following considerations apply to Attendant Serial Calling:

- None

Interactions for Attendant Serial Calling

This section provides information about how the Attendant Serial Calling feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Serial Calling in any feature configuration.

- None

Attendant Split Swap

Use the Attendant Split Swap feature to allow an attendant to alternate between an active call and a split call.

Detailed description of Attendant Split Swap

This section provides a detailed description of the Attendant Split Swap feature.

With the Attendant Split Swap feature, an attendant can alternate between an active call and a split call. To activate this feature, the attendant presses a button on the attendant console. This feature is useful when the attendant must transfer a call, but first must talk independently with each party before completing the transfer.

Hardware requirements for Attendant Split Swap

The Attendant Split Swap feature requires the following hardware:

- An attendant console

Administering Attendant Split Swap

The following steps are part of the administration process for the Attendant Split Swap feature:

- [Assigning a split-swap button](#)

This section describes:

- Any prerequisites for administering the Attendant Split Swap feature
- The screens that you use to administer the Attendant Split Swap feature
- Complete administration procedures for the Attendant Split Swap feature

Prerequisites for administering Attendant Split Swap

You must complete the following actions before you can administer the Attendant Split Swap feature:

- Set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Split Swap

Screen name	Purpose	Fields
<i>Attendant Console</i>	Assign a Split-Swap button.	Any unassigned button in the Feature Button Assignments area.

Assigning a split-swap button

To assign a **split-swap** button:

- 1 Type **change attendant *n***, where *n* is the number of the attendant console. Press **Enter**.
The system displays the *Attendant Console* screen.
- 2 Press **Next** until you see the Feature Button Assignments area ([Figure 28, Attendant Console screen](#), on page 206).

Figure 28: Attendant Console screen

change attendant 1		ATTENDANT CONSOLE		Page 3 of 4
FEATURE BUTTON ASSIGNMENTS				
1: split_____		13: split-swap		
2: dont-split		14: _____		
3: act-tr-grp		15: _____		
4: deact-tr-g		16: _____		
5: _____		17: _____		
6: hold _____ *		18: _____		
7: _____		19: forced-rel		
8: aux-work	RC:	20: _____	Grp:	
9: _____		21: _____		
10: _____		22: _____		
11: _____		23: night-serv *		
12: _____		24: pos-busy__ *		

- 3 In the Feature Button Assignments area, assign **split-swap** to an available button.
In this example, we assign **split-swap** to button **13**.
- 4 Press **Enter** to save your changes.

For more information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Reports for Attendant Split Swap

The following reports provide information about the Attendant Split Swap feature:

- None

Considerations for Attendant Split Swap

This section provides information about how the Attendant Split Swap feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Split Swap under all conditions. The following considerations apply to Attendant Split Swap:

- None

Interactions for Attendant Split Swap

This section provides information about how the Attendant Split Swap feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Split Swap in any feature configuration.

- None

Attendant Timers

Use the Attendant Timers feature to automatically alert the attendant after an administered time interval that calls remain waiting or on hold. The attendant can then reconnect with the caller, and decide whether to terminate the call, or allow the caller to continue waiting.

Detailed description of Attendant Timers

This section provides a detailed description of the Attendant Timers feature.

Attendant Timers automatically alert the attendant after an administered time interval for the following types of calls:

- Extended calls that are waiting to be answered or waiting to be connected to a busy single-line telephone
- One-party calls that are on hold at the console
- Transferred calls that are unanswered after a transfer

The Attendant Timers feature informs the attendant that a call requires additional attention. After the attendant reconnects to the call, the user can either choose to try another extension number, hang up, or continue to wait.

Communication Manager supports a variety of administrable attendant timers. Attendant timers include:

- Unanswered DID Call Timer. Specifies how long a direct inward dialing (DID) call can go unanswered before the system routes the call to the administered DID/TIE/ISDN Intercept Treatment.
- Attendant Return Call Timer. Unanswered calls that the attendant extends return to the same attendant, if the attendant is available. If the same attendant is unavailable, unanswered calls return to the attendant group queue.

The Attendant Return Call Timer is not set for calls that are extended from one attendant to another attendant. A transferred call that times out redirects to an attendant after the interval that is administered for 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 timer expires, the call rings with a secondary, higher-pitch ring. The ringing pattern of a disabled Attendant No Answer Timer does not change over from the primary pattern to the secondary pattern. If the call remains unanswered during this interval, the system routes the call to the attendant group and console where the call was placed in a Position Busy state. This timer does not apply to calls that are placed to the extension of the attendant, or to calls that the attendant originates.

- **Attendant Alerting Interval (Timed Reminder).** Specifies how long a call that terminates at an attendant console can ring with secondary alerting. When the timer expires, the attendant console is placed into position busy mode, and the system forwards the call to the attendant group. If the console where the alerting interval is reached is the last active day console, the system goes into Night Service, if Night Service is enabled. This timer does not apply to calls that are placed to the extension of the attendant, or to calls that the attendant originates.

You can disable the alerting interval. In this case, a call continues to ring at the extension of the original attendant until the caller hangs up, or another feature disconnects the call. If the call reaches the timeout limit for unanswered DID calls during Night Service, for example, the Night Service feature disconnects the call.

- **Line Intercept Tone Timer** — Specifies how long line intercept can be. For example, LITT:10 seconds means that line intercept stops after 10 seconds.

Hardware requirements for Attendant Timers

The Attendant Timers feature requires the following hardware:

- An attendant console

Administering Attendant Timers

The following steps are part of the administration process for the Attendant Timers feature:

- [Setting up Attendant Timers](#)

This section describes:

- Any prerequisites for administering the Attendant Timers feature
- The screens that you use to administer the Attendant Timers feature
- Complete administration procedures for the Attendant Timers feature

Prerequisites for administering Attendant Timers

You must complete the following actions before you can administer the Attendant Timers feature:

- Set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Timers

Screen name	Purpose	Fields
<i>Console Parameters</i>	Set up the timers for the attendant.	All fields in both the Timing and the Incoming Call Reminders areas

Setting up Attendant Timers

To set up Attendant Timers:

- 1 Type **change console-parameters**. Press **Enter**.
The system displays the *Console Parameters* screen.
- 2 Press **Next** until you see the Timing area ([Figure 29, Console Parameters screen](#), on page 211).

Figure 29: Console Parameters screen

change console-parameters	CONSOLE PARAMETERS	Page 2 of 4
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:	

- 3 In the Timing area, complete the following fields:
 - In the Time Reminder on Hold (sec) field, type the number of seconds before a split call returns to the console. Valid entries are a number from 10 to 1024, or blank. A split call can be a call that:
 - The attendant extended to a user and is ringing at the user telephone
 - Otherwise split away from the console

Allow 5 seconds for each ring at all points in a coverage path to ensure that the entire path is completed before the call returns to the console.

- In the **Return Call Timeout (sec)** field, type the number of seconds that a call remains on hold at the console before the system alerts the attendant. In a Centralized Attendant Service (CAS) arrangement, administer the main console and the branch consoles with the same value. Valid entries are a number from 10 to 1024, or blank.
- In the **Time in Queue Warning (sec)** field, type the number of seconds that a call can remain in the attendant queue before the system alerts the attendant. Valid entries are a number from 9 to 999.

4 In the **Incoming Call Reminders** area, complete the following fields:

- In the **No Answer Timeout (sec)** field, type the number of seconds that a call to the attendant can remain unanswered without invoking a more insistent-sounding tone. Valid entries are a number from 10 to 1024, or blank.

Allow 5 seconds for each ring at all points in a coverage path to ensure that the entire path is completed before the call returns to the console.

- In the **Alerting (sec)** field, type the number of seconds after which the system disconnects a held call or an unanswered call from an attendant loop. The system routes the disconnected call either to another attendant or to Night Service. Valid entries are a number from 10 to 1024, or blank.
- In the **Secondary Alert on Held Reminder Calls?** field, perform one of the following actions:
 - Type **y** to start attendant alerting for Held Reminder Calls with secondary alerting
 - Type **n** to have Held Reminder Calls alert the attendant in the same way as normal calls. Normal calls start with primary alerting, and then change to secondary alerting when the No Answer Timeout expires.

5 Press **Enter** to save your changes.

For more information, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Reports for Attendant Timers

The following reports provide information about the Attendant Timers feature:

- None

Considerations for Attendant Timers

This section provides information about how the Attendant Timers feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Timers under all conditions. The following considerations apply to Attendant Timers:

- None

Interactions for Attendant Timers

This section provides information about how the Attendant Timers feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Timers in any feature configuration.

- 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 system redirects the call to an attendant. Redirection occurs even if the call redirects through Call Coverage or Call Forwarding from the transferred-to telephone.

The system redirects an attendant-extended call to coverage instead of returning the call to an attendant if the coverage criteria is met before the Timed Reminder Interval expires. However, the system returns unanswered calls to an attendant at the expiration of the interval.

For any call that alerts an attendant as a coverage call, that is, for any unanswered station-to-station call with the “attd” (attendant) in the **Coverage Path** screen of the called telephone, 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 the call is unanswered.

Attendant Timers

Interactions for Attendant Timers

Attendant Trunk Identification

Use the Attendant Trunk Identification feature to enable the attendant, or a user with a display, to identify a faulty trunk

Detailed description of Attendant Trunk Identification

This section provides a detailed description of the Attendant Trunk Identification feature.

With Attendant Trunk Identification, the attendant, or a user with a display, can identify a faulty trunk. When the attendant or user presses the trk-id button, the system displays the Trunk Access Code (TAC) and trunk member number of a call. The trunk can then be removed from service.

Hardware requirements for Attendant Trunk Identification

The Attendant Trunk Identification feature requires the following hardware:

- An attendant console

Administering Attendant Trunk Identification

This section contains the prerequisites and screens for administering the Attendant Trunk Identification feature.

Prerequisites for administering Attendant Trunk Identification

You must complete the following actions before you can administer the Attendant Trunk Identification feature:

- Set up an attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Attendant Trunk Identification

Screen name	Purpose	Fields
<i>Attendant Console</i>	Administer the trk-id button for the Attendant Trunk Identification feature.	Any available button field in the FEATURE BUTTON ASSIGNMENTS area
<i>Station</i>	Administer the trk-id button for the Attendant Trunk Identification feature.	Any available button field in the FEATURE BUTTON ASSIGNMENTS area

Reports for Attendant Trunk Identification

The following reports provide information about the Attendant Trunk Identification feature:

- None

Considerations for Attendant Trunk Identification

This section provides information about how the Attendant Trunk Identification feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Trunk Identification under all conditions. The following considerations apply to Attendant Trunk Identification:

- None

Interactions for Attendant Trunk Identification

This section provides information about how the Attendant Trunk Identification feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Trunk Identification in any feature configuration.

- None

Audible Message Waiting

Use the Audible Message Waiting feature to alert a user that a message is waiting. The user hears a stutter dial tone when the user goes off hook.

The Audible Message Waiting feature is particularly useful for visually impaired people who cannot see a message waiting light.

Detailed description of Audible Message Waiting

This section provides a detailed description of the Audible Message Waiting feature.

Use the Audible Message Waiting feature to alert a user that a message is waiting. When the user goes off hook, the user hears a stutter tone just before the usual dial tone begins.

Users can access waiting messages from:

- The system memory, where a user can use a display or a voice synthesizer
- A Property Management System (PMS)
- An Avaya Intuity AUDIX voice message system

You must ensure that your users know how to retrieve their messages.

You usually assign Audible Message Waiting on telephones without message waiting lights, such as analog telephones.

If the system loses synchronization between telephones and message-status data, use the Clear Message Waiting Indicators to turn off the message waiting indicators.

Audible Message Waiting requires a separate software right-to-use fee.

Audible Message Waiting might not be applicable in countries that restrict the characteristics of dial tones provided to users.

Hardware requirements for Audible Message Waiting

The Message Waiting feature requires the following hardware:

- None

Administering Audible Message Waiting

The following steps are part of the administration process for the Message Waiting feature:

- [Administering Audible Message Waiting for a user](#)

This section describes:

- Any prerequisites for administering the Message Waiting feature
- The screens that you use to administer the Message Waiting feature
- Complete administration procedures for the Message Waiting feature

Prerequisites for administering Audible Message Waiting

You must complete the following actions before you can administer the Audible Message Waiting feature:

- View the **Optional Features** screen, and ensure that the Audible Message Waiting field is set to **y**. If the Audible Message Waiting field is set to **n**, your system is not enabled for the Audible Message Waiting feature. Contact your Avaya representative before you continue with this procedure.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

Screens for administering Audible Message Waiting

Screen name	Purpose	Fields
Optional Features	Enable Audible Message Waiting for your system.	Audible Message Waiting
Station	Administer Audible Message Waiting for a user.	Audible Message Waiting

Administering Audible Message Waiting for a user

To activate Audible Message Waiting for a user:

- 1 Type **change station n**, where **n** is the extension of the user for whom you want to activate Audible Message Waiting. Press **Enter**.

The system displays the **Station** screen ([Figure 30, Station screen](#), on page 219).

Figure 30: Station screen

FEATURE OPTIONS	STATION
LWC Reception: 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	Restrict Last Appearance? y
Bridged Call Alerting? n	
Active Station Ringing: single	
H.320 Conversion? n	Per Station CPN - Send Calling Number?
Service Link Mode: as-needed	
Multimedia Mode: enhanced	Audible Message Waiting? y
MWI Served User Type:	Display Client Redirection? n
AUDIX Name:	Select Last Used Appearance? n
	Coverage After Forwarding? s
	Multimedia Early Answer? n
	Direct IP-IP Audio Connections? y
	IP Audio Hairpinning? y
Emergency Location Ext: 30019	

2 In the Audible Message Waiting field, perform one of the following actions:

- Type **y**, if you want the user to hear stutter dial tone if a message is waiting when the user goes off hook.
- Type **n**, if you do not want the user to hear stutter dial tone if a message is waiting when the user goes off hook.

The system displays the Audible Message Waiting field only if the Audible Message Waiting field on the **Optional Features** screen is set to **y**.

Note that the Audible Message Waiting field does not control the Message Waiting lamp.

Reports for Audible Message Waiting

The following reports provide information about the Audible Message Waiting feature:

- None

Considerations for Audible Message Waiting

This section provides information about how the Audible Message Waiting feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Audible Message Waiting under all conditions. The following considerations apply to Audible Message Waiting:

- You must tell the user where to call to retrieve messages if the messages are not stored in the system memory for users to access by way of a display or a voice synthesizer.

Interactions for Audible Message Waiting

This section provides information about how the Audible Message Waiting feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Audible Message Waiting in any feature configuration.

- None

Attendant Vectoring

Use the Attendant Vectoring feature to provide attendants with a flexible way to manage incoming calls.

Detailed description of Attendant Vectoring

This section provides a detailed description of the Attendant Vectoring feature.

With the Attendant Vectoring feature, you can establish an attendant vector directory number (VDN), and send attendant group calls through vector processing. This feature is useful when you want flexibility with how the system routes calls when the system is in Night Service mode.

For example, without Attendant Vectoring, calls that are redirected from the attendant console to a night telephone can ring only at that telephone. These calls do not follow a coverage path. With Attendant Vectoring, Night Service calls follow the coverage path of the night telephone. The coverage path can go to another telephone, and eventually to a voice mail system. The caller can then leave a message that can be retrieved at any time.

Attendant Vectoring takes precedence over all local attendant codes that you administer. If Attendant Vectoring is enabled, the system uses call vectors instead of the normal attendant call routing to process attendant-seeking, or dial 0, calls.

For more information about vectors and VDNs, see the “Meet-Me Conference” feature. See also the *Avaya MultiVantage™ Call Center Software Call Vectoring and Expert Agent Selection (EAS) Guide*.

Hardware requirements for Attendant Vectoring

The Attendant Vectoring feature requires the following hardware:

- An attendant console

Administering Attendant Vectoring

The following steps are part of the administration process for the Attendant Vectoring feature:

- [Creating a VDN extension for Attendant Vectoring](#)
- [Assigning the VDN extension for Attendant Vectoring to a console](#)
- [Assigning the VDN extension for Attendant Vectoring to a tenant](#)

This section describes:

- Any prerequisites for administering the Attendant Vectoring feature
- The screens that you use to administer the Attendant Vectoring feature
- Complete administration procedures for the Attendant Vectoring feature

Prerequisites for administering Attendant Vectoring

You must complete the following actions before you can administer the Attendant Vectoring feature:

- Set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.
- On the **Optional Features** screen, verify the values in the following fields. Your license file sets the values in these fields. You cannot manually change these values. If you have any questions, see your Avaya representative for assistance.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

- 1 Click **Next** until you see the Attendant Vectoring field. Ensure that the Attendant Vectoring field is set to **y**.

You cannot use Attendant Vectoring with the Centralized Attendant Service (CAS) feature. Therefore, the Attendant Vectoring field cannot be set to **y** if either the CAS Branch field or the CAS Main field on the **Optional Features** screen is set to **y**.
 - 2 Click **Next** until you see the Tenant Partitioning field. Ensure that the Tenant Partitioning field is set to **n**.
 - 3 Click **Next** until you see the Vectoring (Basic) field and the Vectoring (Prompting) field. Ensure that both these fields are set to **y**.
- On the **Call Vector** screen, verify that the Attendant Vectoring field is set to **y**. To view the **Call Vector** screen, type **change vector n**, where **n** is the number of a vector. Press **Enter**.
 - 1 This field appears on the **Call Vector** screen only if, on the **Optional Features** screen, the Attendant Vectoring field is set to **y**. This field does not appear for S8700 IP-Connect configurations.
 - 2 The Attendant Vectoring field on the **Call Vector** screen defaults to **y** when, on the **Optional Features** screen, the Vectoring (Basic) field and the Vectoring (Prompting) field are set to **y**. You cannot change this setting.
 - 3 The Attendant Vectoring field on the **Call Vector** screen defaults to **n** when, on the **Optional Features** screen, the Vectoring (Basic) field and the Vectoring (Prompting) field are set to **n**. You cannot change this setting.
 - 4 You cannot use Attendant Vectoring with the Meet-me Conference feature. Therefore, the Attendant Vectoring field and the Meet-me Conference field cannot both be set to **y** at the same time.

Screens for administering Attendant Vectoring

Screen name	Purpose	Fields
<i>Optional Features</i>	Ensure that the Attendant Vectoring feature is activated.	Attendant Vectoring
	Ensure that fields are correctly set so that you can: • Verify the Attendant Vectoring feature on the Call Vector screen • Administer the Attendant Vectoring feature on the Console Parameters screen. See the “Prerequisites” section for the correct settings.	• Tenant Partitioning • Vectoring (Basic) • Vectoring (Prompting)
<i>Call Vector</i>	Ensure that the Attendant Vectoring field is set to y .	Attendant Vectoring
<i>Vector Directory Number</i>	Create an Attendant Vectoring VDN extension.	Attendant Vectoring
<i>Console Parameters</i>	Assign the Attendant Vectoring VDN extension to a console.	Attendant Vectoring VDN
<i>Tenant</i>	Assign the Attendant Vectoring VDN extension to a tenant.	Attendant Vectoring VDN

Creating a VDN extension for Attendant Vectoring

Use the **Vector Directory Number** screen to define vector directory numbers (VDNs) for the Call Vectoring feature. A VDN is an extension that gives you access to a call vector. Each VDN is mapped to one call vector.

VDNs are not assigned to physical equipment. You gain access to a VDN through direct dial central office (CO) trunks that are mapped to the VDN, direct inward dial (DID) trunks, and listed directory number (LDN) calls. The VDN can be a night destination for LDN.

To create a VDN extension for Attendant Vectoring:

- 1 Type **change vdn *n***, where *n* is the VDN extension. Press **Enter**.

The system displays the **Vector Directory Number** screen ([Figure 31, Vector Directory Number screen](#), on page 224).

Figure 31: Vector Directory Number screen

change vdn 5000	VECTOR DIRECTORY NUMBER	Page 1 of 1
	Extension: 5000	
	Name:	
	Vector Number: 234	
	Attendant Vectoring: y	
	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:	

2 In the Attendant Vectoring field, perform one of the following actions:

- If you want this VDN to be an attendant vector, type **y**.
- If you do not want this VDN to be an attendant vector, type **n**.

You cannot use Attendant Vectoring with the Meet-me Conference feature. Therefore, the Attendant Vectoring field and the Meet-Me Conferencing field cannot both be set to **y** at the same time.

3 Press **Enter** to save your changes.

Assigning the VDN extension for Attendant Vectoring to a console

The Attendant Vectoring VDN field appears on the *Console Parameters* screen only if, on the *Optional Features* screen, the:

- Attendant Vectoring field is set to **y**
- Tenant Partitioning field is set to **n**

To assign the VDN extension for Attendant Vectoring to a console:

1 Type **change console-parameters n**, where **n** is the assigned number of the attendant console. Press **Enter**.

The system displays the *Console Parameters* screen ([Figure 32, Console Parameters screen](#), on page 225).

Figure 32: Console Parameters screen

change console-parameters CONSOLE PARAMETERS Attendant Group Name: OPERATOR COS: 0 Calls in Queue Warning: 5 Ext Alert Port (TAAS): CAS: none IAS (Branch)? n IAS Att. Access Code: Backup Alerting? n Attendant Vectoring VDN: 5000	Page 1 of 4 COR: 0 Attendant Lockout? y Night Service Act. Ext.: IAS Tie Trunk Group No.: Alternate FRL Station: DID-LDN Only to LDN Night Ext? n

- 2 In the Attendant Vectoring VDN field, type the VDN extension that you have created for Attendant Vectoring.
- 3 Press **Enter** to save your changes.

Assigning the VDN extension for Attendant Vectoring to a tenant

Use the **Tenant** screen to define tenants that access the system. If your server uses more than one tenant, see the “Tenant Partitioning” feature for more information.

To assign the VDN extension for Attendant Vectoring to a tenant:

- 1 Type **change tenant *n***, where *n* is the assigned number of the tenant. Press **Enter**.
The system displays the **Tenant** screen ([Figure 33, Tenant screen](#), on page 225).

Figure 33: Tenant screen

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: 5000 Emergency Number:	

- 2 In the Attendant Vectoring VDN field, type the VDN extension that you have created for Attendant Vectoring.
- 3 Press **Enter** to save your changes.

For more information, [click here](#), or see the *Administrator’s Guide for Avaya Communication Manager*.

Reports for Attendant Vectoring

The following reports provide information about the Attendant Vectoring feature:

- None

Considerations for Attendant Vectoring

This section provides information about how the Attendant Vectoring feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Attendant Vectoring under all conditions. The following considerations apply to Attendant Vectoring:

- Teletypewriter device (TTY) for the hearing impaired

Unlike fax machines and modems, a TTY has no handshake tone and no carrier tone. A TTY is silent when not transmitting. Systems cannot automatically identify TTY callers.

However, the absence of these special tones also means that voice and TTY tones can be intermixed in prerecorded announcements. The ability to provide a hybrid voice-and-TTY announcement, when combined with the Attendant Vectoring capability, can permit a single telephone number to accommodate both voice and TTY callers.

Interactions for Attendant Vectoring

This section provides information about how the Attendant Vectoring feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Attendant Vectoring in any feature configuration.

- Centralized Attendant Service (CAS)

You cannot use Attendant Vectoring with the Centralized Attendant Service (CAS) feature. Therefore, the **Attendant Vectoring** field cannot be set to **y** if either the **CAS Branch** field or the **CAS Main** field on the **Optional Features** screen is set to **y**.

- Meet-me Conference

You cannot use Attendant Vectoring with the Meet-me Conference feature. Therefore, the **Attendant Vectoring** field and the **Meet-me Conference** field cannot both be set to **y** at the same time.

AUDIX One-Step Recording

Use the AUDIX One-Step Recording feature to record telephone conversations by pressing a single button. AUDIX One-Step Recording then stores the recorded conversation as a message in the voice mailbox of the user.

Detailed description of AUDIX One-Step Recording

This section provides a detailed description of the AUDIX One-Step Recording feature.

The AUDIX One-Step Recording feature is available with Avaya Communication Manager release 1.3 (V11) or later.

- Communication Manager release 1.3 supports AUDIX One-Step Recording only when the AUDIX system is local.
- Communication Manager release 2.0 and later supports AUDIX One-Step Recording with a local AUDIX system and with a remote AUDIX system. To use a remote AUDIX system, the person recording the conversation must have Communication Manager release 2.0 or later.

The AUDIX One-Step Recording feature uses AUDIX to record a telephone conversation. A user needs to press only one feature button on the telephone to activate this feature. AUDIX One-Step Recording then stores the recorded conversation as a message in the voice mailbox of the user. The system allows AUDIX One-Step Recording only after a call is answered.

NOTE:

Some countries, states, and localities have laws that determine if and under what circumstances you can record telephone conversations. Before you administer the AUDIX One-Step Recording feature, you must understand and comply with these laws.

Feature button

The AUDIX One-Step Recording feature uses an administered feature button, **audix-rec**. The administrator uses the **Station** screen to assign this button for each telephone. When you assign the feature button, the system requires you to provide the extension for the AUDIX hunt group of the user. Users cannot access AUDIX One-Step Recording on any telephone that does not have an administrable feature button, or on an attendant console.

Language options

Five languages are available for the feature button labels for AUDIX One-Step Recording:

- English
- Italian
- French
- Spanish
- A user-defined language

The feature button labels for AUDIX One-Step Recording are available in English, with predefined Italian, French, and Spanish translations. The administrator cannot change the text of the English, the Italian, the French, or the Spanish feature button labels.

The administrator can translate the feature button labels into a user-defined language. This translation can be any other language that the customer chooses, such as German. The administrator can use only one user-defined language throughout the system.

Periodic alerting tone

While AUDIX records the conversation, all parties on the call might hear a periodic alerting tone. This alerting tone reminds all parties on the call that AUDIX is recording the conversation. The tone that plays is a zip tone.

You choose the time interval to play the periodic alerting tone. If you set the time interval or the periodic alerting tone to zero, the parties hear no alerting tone.

Ready indication tone

When AUDIX starts recording the conversation, the system plays a ready indication tone. The tone that plays is a zip tone.

The administrator can set the ready indication tone to play to:

- All the parties on the call
- The initiator only
The “initiator” is the user who activates the AUDIX One-Step Recording feature.
- None of the parties on the call

Recording delay timer

When the user presses the **audix-rec** feature button, AUDIX answers. After AUDIX answers, the system uses the recording delay timer to wait for AUDIX to get ready to interpret digits. The system then sends digit **1** to tell AUDIX to start recording.

The recording delay timer is preset to 500 milliseconds. This delay is sufficient for most recorded conversations. If you have to change this setting, see [Assigning AUDIX One-Step Recording parameters](#) on page 230.

Hardware requirements for AUDIX One-Step Recording

The AUDIX One-Step Recording feature requires the following hardware:

- A system with one of the following:
 - DEFINITY® AUDIX
 - Intuity® AUDIX, R4 or later (CLAN integration or DCUI/X.25 integration)
 - IA770, R1.1 or later
- A digital telephone or an IP telephone with administrable feature buttons

Administering AUDIX One-Step Recording

The following steps are part of the administration process for the AUDIX One-Step Recording feature:

- [Assigning AUDIX One-Step Recording parameters](#)
- [Translating telephone feature buttons and labels](#)
- [Assigning the feature button to a telephone](#)
- [Changing the characteristics of the zip tone](#)

This section describes:

- Any prerequisites for administering the AUDIX One-Step Recording feature
- The screens that you use to administer the AUDIX One-Step Recording feature
- Complete administration procedures for the AUDIX One-Step Recording feature

Prerequisites for administering AUDIX One-Step Recording

You must complete the following actions before you can administer the AUDIX One-Step Recording feature:

- On the **Optional Features** screen, ensure that the G3 Version field is set to **V11** or later. If this field is not set to **V11** or later, your system is not enabled for the AUDIX One-Step Recording feature. Contact your Avaya representative for assistance.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

For a complete description of the many **Optional Features** screens, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering AUDIX One-Step Recording

Screen name	Purpose	Fields
Optional Features	Ensure that you have Communication Manager version 1.3 (V11) or later.	G3 Version
Feature-Related System Parameters	Set the time interval, in milliseconds, for the recording delay timer.	Recording Delay Timer (msec)
	Assign who on the call can hear the ready indication tone.	Apply Ready Indication Tone To Which Parties In The Call
	Set the time interval, in seconds, for the periodic alerting tone.	Interval For Applying Periodic Alerting Tone (seconds)
Language Translations	If needed, change the translation of the audix-rec feature button to a user-defined language.	Audix Recording
	If needed, change the translation of the Audix Record button label to a user-defined language.	Audix Record
Station	Assign the audix-rec feature button, and associate the hunt group extension of the user.	Button Assignments

Assigning AUDIX One-Step Recording parameters

To assign the recording delay timer, the ready indication tone, and the periodic alerting tone:

- 1 Type **change system-parameters features**. Press **Enter**.
The system displays the **Feature-Related System Parameters** screen.
- 2 Click **Next** until you see the AUDIX One-step Recording area ([Figure 34, Feature-Related System Parameters screen](#), on page 231).

Figure 34: Feature-Related System Parameters screen

change system-parameters features	Page 6 of 12
FEATURE-RELATED SYSTEM PARAMETERS	
CONFERENCE/TRANSFER	
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? n	
AUDIX ONE-STEP RECORDING	
Recording Delay Timer (msec): 500	
Apply Ready Indication Tone To Which Parties In The Call? all	
Interval For Applying Periodic Alerting Tone (seconds): 15	

- 3 If you need to change the time between when AUDIX answers and when AUDIX starts recording, change the value in the **Recording Delay Timer (msec)** field. The time is measured in milliseconds in increments of 100. The default setting is 500 milliseconds.

After AUDIX starts recording, Communication Manager must filter out the AUDIX tone. The user must wait longer than the Recording Delay Timer setting before the user notices that recording has started.
- 4 In the **Apply Ready Indication Tone To Which Parties In The Call** field, type one of the following values. These values indicate who hears the ready indication tone:
 - If you want all parties on the call to hear the ready indication tone, leave the default set to **all**. The screen displays the **Interval For Applying Periodic Alerting Tone (seconds)** field.
 - If you want only the initiator to hear the ready indication tone, right-click the field and select **initiator**. The screen does not display the **Interval For Applying Periodic Alerting Tone (seconds)** field.
 - If you want no one on the call to hear the ready indication tone, right-click the field and select **none**. The screen does not display the **Interval For Applying Periodic Alerting Tone (seconds)** field.
- 5 In the **Interval For Applying Periodic Alerting Tone (seconds)** field, type a value between **0** and **60** seconds. This value indicates how often all parties on the call hear the periodic alerting tone.

The default value is **15** seconds. If you set the value to **0**, no one on the call hears the periodic alerting tone.
- 6 Press **Enter** to save your changes.

Translating telephone feature buttons and labels

To translate telephone feature buttons and labels for AUDIX One-Step Recording to a user-defined language, you must complete the following procedures:

- [Translating the AUDIX One-Step Recording feature display to a user-defined language](#)
- [Translating the AUDIX One-Step Recording button label to a user-defined language](#)

Translating the AUDIX One-Step Recording feature display to a user-defined language

To translate the text that appears on the telephone display when this feature is recording to a user-defined language:

- 1 Type **change display-messages view-buttons**. Press **Enter**.
The system displays the *Language Translations* screen.
- 2 Click **Next** until you see the Audix Recording field ([Figure 35, Language Translations screen](#), on page 232).

Figure 35: Language Translations screen

change display-messages view-buttons		Page 9 of 9
LANGUAGE TRANSLATIONS		
English	Translation	
1. Station Lock	*****	
2. License Error	*****	
3. Conference Display	*****	
4. Conf/Trans Toggle-Swap	*****	
5. Posted Messages	*****	
6. Audix Recording	*****	

- 3 In the Translation field, type a translated name for Audix Recording into the user-defined language.
Note that the language translations for the English, the Italian, the French, and the Spanish feature display are predefined. You cannot change the text for these translations.
- 4 Press **Enter** to save your changes.

Translating the AUDIX One-Step Recording button label to a user-defined language

The 2420 DCP telephone and the 4620 IP telephone have digital button labels instead of paper labels. If you need to translate the AUDIX One-Step Recording button labels into a user-defined language for these telephones, follow this procedure.

To translate the AUDIX One-Step Recording button label for the 2420 DCP telephone and the 4620 IP telephone to a user-defined language:

- 1** Type **change display-messages button-labels**. Press **Enter**.
The system displays the *Language Translations* screen.
- 2** Click **Next** until you see the Audix Record field ([Figure 36, Language Translations screen](#), on page 233).

Figure 36: Language Translations screen

change display-messages button-labels
Page 9 of 9

LANGUAGE TRANSLATIONS

English	Translation
1. Alternate FRL	1. *****
2. ANI Request	2. *****
3. Assist	3. *****
4. ASVN Halt	4. *****
5. AttQueueCall	5. *****
6. AttQueueTime	6. *****
7. Audix Record	7. *****
8. Auto Callback	8. *****
9. Auto Chk Halt	9. *****
10. AutoIC	10. *****
11. Auto In	11. *****
12. AutoWakeAlarm	12. *****
13. Auto Wakeup	13. *****
14. AuxWork	14. *****
15. Busy	15. *****

- 3** In the Translation field, type a translated name for the Audix Record button label into the user-defined language.

Note that the language translations for the English, the Italian, the French, and the Spanish feature buttons are predefined. You cannot change the text for these translations.
- 4** Press **Enter** to save your changes.

Assigning the feature button to a telephone

To assign the **audix-rec** feature button to an individual telephone:

- 1 Type **change station *n***, where *n* is the extension of the telephone. Press **Enter**.
The system displays the **Station** screen.
- 2 Click **Next** until you see the Button Assignments area ([Figure 37, Station screen](#), on page 234).

Figure 37: Station screen

change station 9876		Page 3 of 3	
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. audix-rec Ext: 9800		9.	
5.		10.	

- 3 Right-click a button field that is not assigned to see a list of button names. Select **audix-rec** from the list.
The screen displays an **Ext** field.
- 4 In the **Ext** field, type the AUDIX hunt group extension of the user.
- 5 Press **Enter** to save your changes.

Changing the characteristics of the zip tone

When the AUDIX One-Step Recording feature is recording a conversation, the tone that a user hears is a zip tone. By default, a zip tone plays at a frequency of 480 Hz for 500 milliseconds, followed by silence.

If you need to change the characteristics of the zip tone, the screen you use depends on what version of Communication Manager you have.

To change the characteristics of the zip tone if you have Communication Manager release 2.0 (V12) or later:

- 1 Type **change tone-generation**. Press **Enter**.

The system displays the **Tone Generation** screen.

- 2 Click **Next** until you come to:

- an existing **Tone Generation Customized Tones** screen for the zip tone
If a screen already exists for the zip tone, the word **zip** appears in the **Tone Name** field.
- a blank **Tone Generation Customized Tones** screen ([Figure 38, Tone Generation Customized Tones screen](#), on page 235).

Figure 38: Tone Generation Customized Tones screen

change tone-generation
Page 4 of 21

TONE GENERATION CUSTOMIZED TONES

Tone Name	Cadence Step	Tone (Frequency/Level)
	1:	
	2:	
	3:	
	4:	
	5:	
	6:	
	7:	
	8:	
	9:	
	10:	
	11:	
	12:	
	13:	
	14:	
	15:	

If a screen already exists for the zip tone, skip the next step and go to step #4.

- 3 Right-click the **Tone Name** field to see a list of options. Select **zip** from the list.
- 4 Right-click the **Cadence Step 1** field to see a list of options. Select a tone frequency and level combination from the list.
The screen displays a **Duration (msec)** field.
- 5 In the **Duration (msec)** field, type the number of milliseconds you want the zip tone to play.
- 6 Right-click the **Cadence Step 2** field to see a list of options. Select **silence** from the list.
The screen displays a **Duration (msec)** field.
- 7 In the **Duration (msec)** field, type the number of milliseconds you want the zip tone to remain silent after the system plays the tone.
- 8 Press **Enter** to save your changes.

To change the characteristics of the zip tone if you have Communication Manager, release 1.3 (V11) or earlier:

- 1 If you are running Communication Manager, release 1.3 (V11) or earlier, type **change system-parameters country-options**. Press **Enter**.

The system displays the *System Parameters Country-Options* screen.

- 2 Click **Next** until you come to:
 - an existing *System Parameters Country-Options* screen for the zip tone
If a screen already exists for the zip tone, the word **zip** appears in the Tone Name field.
 - a blank *System Parameters Country-Options* screen ([Figure 39, System Parameters Country-Options screen](#), on page 236).

Figure 39: System Parameters Country-Options screen

change system-parameters country-options		Page 7 of 24
SYSTEM PARAMETERS COUNTRY-OPTIONS		
Tone Name	Cadence Step	Tone (Frequency/Level)
	1:	
	2:	
	3:	
	4:	
	5:	
	6:	
	7:	
	8:	
	9:	
	10:	
	11:	
	12:	
	13:	
	14:	
	15:	

If a screen already exists for the zip tone, skip the next step and go to step #4.

- 3 Right-click the Tone Name field to see a list of options. Select **zip** from the list.
- 4 Right-click the Cadence Step 1 field to see a list of options. Select a tone frequency and level combination from the list.
The screen displays a Duration (msec) field.
- 5 In the Duration (msec) field, type the number of milliseconds you want the zip tone to play.
- 6 Right-click the Cadence Step 2 field to see a list of options. Select **silence** from the list.
The screen displays a Duration (msec) field.
- 7 In the Duration (msec) field, type the number of milliseconds you want the zip tone to remain silent after the system plays the tone.
- 8 Press **Enter** to save your changes.

End-user procedures for AUDIX One-Step Recording

End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for the AUDIX One-Step Recording feature:

A user announces to the other parties on the call that he or she wants to record the conversation. This user, whom we now call the “initiator”, gets permission from the parties and presses the **audix-rec** button to record the conversation.

When the initiator presses the **audix-rec** button, the LED for the **audix-rec** button flashes. After a few seconds, the telephone displays of all internal users on the call change to CONFERENCE. The number of parties on the call increases by 1. The LED on the telephone of the initiator stays on and no longer flashes. This light indicates that AUDIX is ready to record.

AUDIX starts to record the conversation. The system plays the ready indication tone to indicate that recording is started. Only the initiator hears any AUDIX announcements.

To stop the recording at any time, the initiator can press the **audix-rec** button again. The LED on the telephone of the initiator goes out. The number of parties on the call decreases by 1. The call remains active. The initiator can press the **audix-rec** button to start and stop recording the same conversation any number of times. Each time creates a separate recorded message.

If the initiator hangs up while AUDIX is recording the conversation, the recording ends. If the call is originally a two-party call and the other party hangs up, the recording ends. If the call is originally a multiple-party conference call and someone other than the initiator hangs up, the recording continues.

When the recording ends, the system saves the recorded conversation in the voice mailbox of the initiator as a new voice mail message.

Reports for AUDIX One-Step Recording

The following reports provide information about the AUDIX One-Step Recording feature:

- None

Considerations for AUDIX One-Step Recording

This section provides information about how the AUDIX One-Step Recording feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of AUDIX One-Step Recording under all conditions. The following considerations apply to the AUDIX One-Step Recording feature:

- Capacity constraints and feature limitations

AUDIX One-Step Recording has the following capacity constraints and feature limitations:

- Only one simultaneous AUDIX recording is allowed for each call.
- The **audix-rec** feature button works as a toggle button. When the button is active, the button is associated with only one call. A user cannot use the **audix-rec** feature button on two calls simultaneously.
- Attendant consoles do not have an **audix-rec** feature button. Attendants cannot use the AUDIX One-Step Recording feature.
- Service observers who are actively involved in service observing cannot use the AUDIX One-Step Recording feature.

- Denial scenarios

The AUDIX One-Step Recording feature is denied, and the feature button flutters, if a user presses the **audix-rec** button when:

- No call is active on the telephone
- The user is still dialing digits
- The connection to AUDIX is not operating
- All AUDIX ports are busy
- The number of parties on the call reaches the administered maximum
- Another party on the call is already recording the conversation with AUDIX One-Step Recording
- The user starts the recording from a bridged call appearance
- Service observers are actively involved in service observing
- The incoming or outgoing call is ringing and is not answered

- Security

Some countries, states, and localities have laws that determine if and under what circumstances you can record telephone conversations. Before you administer the AUDIX One-Step Recording feature, you must understand and comply with these laws.

- Serviceability

This feature depends on AUDIX to function. The administrator is responsible for properly administering AUDIX. AUDIX must work for the telephone user before the administrator applies this feature to the telephone. Specifically, the administrator needs to set the appropriate **Mailbox Size** and **Voice Mail Message Maximum Length** on AUDIX for the user.

The **audix-rec** button requires the extension of the AUDIX hunt group of the user. The administrator must type the correct extension on the **Station** screen.

Interactions for AUDIX One-Step Recording

This section provides information about how the AUDIX One-Step Recording feature interacts with other features in your system. Use this information to ensure that you receive the maximum benefits of AUDIX One-Step Recording in any feature configuration. The following interactions apply to the AUDIX One-Step Recording feature:

- Bridged Call Appearance
 - A user cannot activate or deactivate the **audix-rec** feature button from a bridged call appearance.
 - If the administrator sets the ready indication tone to play to all parties, parties on bridged call appearances hear the tone. If the administrator sets the ready indication tone to play to the initiator only, parties on bridged call appearances do not hear the tone.
- Call Park

When AUDIX One-Step Recording is active on a call, the system does not allow the user to park the call. If the user attempts to park the call, the call appearance LED flutters.

When a call is parked, the system does not allow the user to activate AUDIX One-Step Recording. If the user attempts to activate AUDIX One-Step Recording, the feature button LED flutters.
- Conference

When the recording request is in process, the LED is flashing. If anyone attempts to conference in another party, AUDIX One-Step Recording stops. The LED goes out.

When the recording starts, the LED is on but not flashing. Any party on the call can conference in another party. The recording continues.

If AUDIX One-Step Recording is in progress on two separate calls, the system does not allow the user to conference the two calls together.
- Coverage Answer Group

An answer group contains up to eight members who act as a coverage point for another user. For example, if several attendants are responsible to answer redirected calls from a department, assign all attendants to an answer group. The administrator assigns a group number to the answer group. That group number appears in the coverage path of that department. All telephones in an answer group ring simultaneously. Any member of the group can answer the call.

When a member of a Coverage Answer Group answers a call, any of the parties on the call can press the **audix-rec** button to record the conversation. The system stores the recorded message in the voice mailbox of the party who answers the call.
- Drop Last

After AUDIX starts to record, the initiator becomes the control party. If the initiator presses the **Drop** button, AUDIX One-Step Recording stops and the LED goes out. If other parties on the call press the **Drop** button, the system ignores the request.
- Group Paging that uses a speakerphone

With Group Paging, users can make an announcement over a group of digital speakerphones. Neither the group members who receive a page, nor the originator of the page, can use AUDIX One-Step Recording to record the conversation. If anyone on the call presses the **audix-rec** button, the system ignores the request. The LED flutters.

- **Hunt Group**

If a member of a hunt group answers a call, the parties on the call can use AUDIX One-Step Recording to record the conversation. The system stores the recorded message in the voice mailbox of the party who answers the call.
- **Hunt Group queueing**

The system does not allow the user to activate AUDIX One-Step Recording when the following conditions apply.

 - Queue is enabled for the AUDIX hunt group
 - All the AUDIX ports are busy
- **Meet-me Conference**

Parties on a Meet-me Conference can use AUDIX One-Step Recording. Conference parties can selectively use the Display and Drop feature to drop the AUDIX One-Step Recording hunt group extension from the call. The LED goes out.
- **Mode Code interface**

Voice mail connections through the Mode Code interface rely on digits that are sent over analog ports. The AUDIX One-Step Recording feature does not work with a Mode Code connection.
- **Terminating Extension Group (TEG)**

Neither group members who receive TEG calls, nor the originators of TEG calls, can use AUDIX One-Step Recording to record the conversation. If anyone on the call presses the **audix-rec** button, the system ignores the request. The LED flutters.
- **Transfer**

When the recording request is in process, the LED is flashing. If anyone attempts to transfer the call to another party, AUDIX One-Step Recording stops. The LED goes out.

When the recording starts, the LED is on but not flashing. Any party on the call can transfer the call to another party.

 - If the initiator transfers the call, recording automatically stops after the transfer.
 - If another party on the call transfers the call, recording continues.

If AUDIX One-Step Recording is in progress on two separate calls, the system does not allow a user to transfer one call to the other.
- **Vector Directory Number (VDN)**

When a call is answered through a VDN, any of the parties on the call can press the **audix-rec** button to record the conversation.
- **Whisper Page**

With Whisper Page, no parties on the call can use AUDIX One-Step Recording to record the conversation. If anyone on the call presses the **audix-rec** button, the system ignores the request. The LED flutters.

Troubleshooting AUDIX One-Step Recording

This section lists the known or common problems that users might experience with the AUDIX One-Step Recording feature.

Problem	Possible cause	Action
The user does not have the audix-rec button on the telephone.	The audix-rec button is not assigned to the telephone of the user.	Use the change station <i>n</i> command, where <i>n</i> is the extension of the telephone, to add the audix-rec button. For more information, see Assigning the feature button to a telephone on page 234.
	The telephone of the user does not support buttons that can be administered.	This user cannot access the AUDIX One-Step Recording feature.
Nothing happens when the user presses the audix-rec button. The button light flutters.	The user was not on an active call.	Inform the user that the audix-rec button only works if the user is on an active call.
	The connection to AUDIX is not operating.	Ensure that AUDIX is operating correctly. If not, the user must wait for the system to reestablish the connection.
	The LAN connection between the AUDIX server and the Communication Manager server is not operating.	Ensure that AUDIX is operating correctly. If not, the user must wait for the system to reestablish the connection.
	All the AUDIX ports are busy.	The user must wait for an AUDIX port to be released.
	The number of parties on the call reached the administered maximum.	[<i>Note to reviewers: Please indicate what the administrator must do to increase the maximum number of people on a conference call.</i>]
	Another person on the call is already using the AUDIX One-Step Recording feature to record the conversation.	The user cannot record the conversation if someone else on the call is already using this feature.
	The user attempted to start the AUDIX One-Step Recording feature from a bridged call appearance.	The user cannot record the conversation from a bridged call appearance.
	Remote AUDIX is configured for Communication Manager release 1.3.	• Move the voice mail service of the user to a local AUDIX. • Upgrade the user to Communication Manager release 2.0 or later.

Problem	Possible cause	Action
The user hears a beep every few seconds during recording.	The alerting tone interval is set too low.	Set the alerting tone interval to a number between 15 and 60. For more information, see Assigning AUDIX One-Step Recording parameters on page 230.
The user does not hear a beep during recording.	The ready indication tone is set to play to only the initiator, or to no one.	Set the ready indication tone to all . For more information, see Assigning AUDIX One-Step Recording parameters on page 230.
	The alerting tone interval is set to zero.	Set the alerting tone interval to a number between 15 and 60. For more information, see Assigning AUDIX One-Step Recording parameters on page 230.
	The zip tone is set to silence instead of to a frequency and duration.	Reset the characteristics of the zip tone. For more information, see Changing the characteristics of the zip tone on page 234.
After the green LED is steady and the recording starts, the user hears an AUDIX announcement.	If configured with Intuity [®] AUDIX, ARIA TUI might be turned on. The Recording Delay Timer is not set high enough.	Set the Recording Delay Timer to a higher number. For more information, see Assigning AUDIX One-Step Recording parameters on page 230.

Automated Attendant

Use the Automated Attendant feature to allow callers to dial an extension without the need for an attendant to connect the call.

Detailed description of Automated Attendant

This section provides a detailed description of the Automated Attendant feature.

A caller dials any extension on the system. The Automated Attendant feature uses vectors to route the call to that extension. This feature reduces the need for live attendants, and this can help to reduce costs.

The Automated Attendant feature works together with the Attendant Vectoring feature. For more information, see the “Attendant Vectoring” feature. For more information about vectors and vector directory numbers (VDNs), see the “Meet-Me Conference” feature.

The *Avaya MultiVantage™ Call Center Software Call Vectoring and Expert Agent Selection (EAS) Guide* contains a detailed description of Automated Attendant. This guide also gives a sample vector that you can use for Automated Attendant. The guide contains essential information about how to use the Automated Attendant feature.

Hardware requirements for Automated Attendant

The Automated Attendant feature requires the following hardware:

- An attendant console

Administering Automated Attendant

The following steps are part of the administration process for the Automated Attendant feature:

- [Setting the prompting timeout](#)
- [Administering a vector directory number](#)
- [Administering announcements](#)
- [Controlling hunt groups by vector](#)
- [Administering call vectors](#)
- [Assigning a caller information button on a multiappearance telephone](#)
- [Assigning a caller information button on an attendant console](#)

This section describes:

- Any prerequisites for administering the Automated Attendant feature
- The screens that you use to administer the Automated Attendant feature
- Complete administration procedures for the Automated Attendant feature

Prerequisites for administering Automated Attendant

You must complete the following actions before you can administer the Automated Attendant feature:

- Set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.
- On the **Optional Features** screen, ensure that the **Vectoring (Prompting)** field is set to **y**. Your license file sets this value. You cannot manually change these values. If you have any questions, see your Avaya representative for assistance.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

Screens for administering Automated Attendant

Screen name	Purpose	Fields
Optional Feature	Ensure that the Automated Attendant feature is activated.	Vectoring (Prompting)
Feature-Related System Parameters	Set the prompting timeout period.	Prompting Timeout
Vector Directory Number	Assign a VDN to an extension.	All
Announcements/Audio Sources	Complete all fields for each extension that provides an Automated Attendant announcement.	All
Hunt Group	Indicate that the system controls hunt groups with vectors.	Vector
Call Vector	Complete a Call Vector screen for each Automated Attendant vector.	All
Station	Assign a callr-info display button for multiappearance telephones.	Any blank field in either the BUTTON ASSIGNMENTS area or the FEATURE BUTTON ASSIGNMENTS area
Attendant Console	Assign a callr-info display button for attendant consoles.	Any blank field in the FEATURE BUTTON ASSIGNMENTS area

Setting the prompting timeout

To set the prompting timeout:

- 1 Type **change system-parameters features**. Press **Enter**.
The system displays the *Feature-Related System Parameters* screen.
- 2 Click **Next** until you see the VECTORING area ([Figure 40, Feature-Related System Parameters screen](#), on page 245).

Figure 40: Feature-Related System Parameters screen

change system-parameters features	Page 7 of 8
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	
SERVICE OBSERVING	
Service Observing Warning Tone? n	
ASAI	
Call Classification After Answer Supervision? n	Send UCID to ASAI? n

- 3 In the Prompting Timeout (sec) field, type the number of seconds before the **collect digits** command times out for callers who use rotary dialing. This value must be a number from 4 to 10. The default is 10.

The Prompting Timeout (sec) field appears only if the Vectoring (Prompting) field on the *Optional Features* screen is set to **y**.
- 4 Press **Enter** to save your changes.

Administering a vector directory number

To administer a vector directory number (VDN), see the *Avaya MultiVantage™ Call Center Software Call Vectoring and Expert Agent Selection (EAS) Guide*.

See also the “Attendant Vectoring” feature.

Administering announcements

To administer announcements for the Automated Attendant, see the “Announcements” feature.

Controlling hunt groups by vector

To control hunt groups by vector:

- 1 Type **change hunt-group *n***, where *n* is the number of the hunt group that you want to control by vector. Press **Enter**.

The system displays the **Hunt Group** screen ([Figure 41, Hunt Group screen](#), on page 246).

Figure 41: Hunt Group screen

change hunt-group 12345		Page 1 of 3	
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 Disp: _____			
Queue Length: ____			
Calls Warning Threshold: ____	Port: x	Extension: ____	
Time Warning Threshold: ____	Port: x	Extension: ____	

- 2 In the **Vector** field, type **y** to indicate that this hunt group is vector controlled.
You can change the **Vector** field to **y** only if the **Vectoring (Basic)** field on the **Optional Features** screen is set to **y**. For more information on setting up hunt groups, see the “Hunt Group” feature.
- 3 Press **Enter** to save your changes.

Administering call vectors

To administer call vectors for each Automated Attendant vector, see the “Call Vector” feature.

Assigning a caller information button on a multiappearance telephone

You can administer any multiappearance display telephone to have a caller information (**callr-info**) button. This button displays digits that were collected for the last **collect digits** command.

To assign a **callr-info** button on a multiappearance telephone:

- 1 Type **change station *n***, where *n* is the extension of the telephone that you want to change. Press **Enter**.

The system displays the **Station** screen.

- 2 Press **Next** until you see the BUTTON ASSIGNMENTS area ([Figure 42, Station screen](#), on page 247).
If all the buttons are assigned in this area, press **Next** until you see the FEATURE BUTTON ASSIGNMENTS area.

Figure 42: Station screen

change station 60006	STATION	Page 3 of 5
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: cpn-blk	
2: call-appr	5: cpn-unblk	
3: call-appr	6: callr-info	

- 3 In the BUTTON ASSIGNMENTS area or in the FEATURE BUTTON ASSIGNMENTS area, assign **callr-info** to an available button.
In this example, we assign **callr-info** to button **6** in the BUTTON ASSIGNMENTS area.
- 4 Press **Enter** to save your changes.

Assigning a caller information button on an attendant console

You can administer an attendant console to have a caller information (**callr-info**) button. This button displays digits that were collected for the last **collect digits** command.

To assign a **callr-info** button on an attendant console:

- 1 Type **change attendant n**, where **n** is the number of the attendant console to which you want to assign a **callr-info** button. Press **Enter**.
The system displays the *Attendant Console* screen.
- 2 Press **Next** until you see the Feature Button Assignments area ([Figure 43, Attendant Console screen](#), on page 248).

Figure 43: Attendant Console screen

change attendant 1		ATTENDANT CONSOLE		Page 3 of 4
FEATURE BUTTON ASSIGNMENTS				
1: split_____			13: callr-info	
2: dont-split			14: _____	
3: act-tr-grp			15: _____	
4: deact-tr-g			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__ *	

- 3 In the FEATURE BUTTON ASSIGNMENTS area, assign **callr-info** to an available button.
In this example, we assign **callr-info** to button **13**.
- 4 Press **Enter** to save your changes.

For more information, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Reports for Automated Attendant

The following reports provide information about the Automated Attendant feature:

- None

Considerations for Automated Attendant

This section provides information about how the Automated Attendant feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Automated Attendant under all conditions. The following considerations apply to Automated Attendant:

- Automated Attendant competes with several features for ports on the call classifier - detector circuit pack or an equivalent circuit pack.

Interactions for Automated Attendant

This section provides information about how the Automated Attendant feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Automated Attendant in any feature configuration.

- Avaya INTUITY AUDIX

Automated Attendant gives the caller the option to leave a message or wait in a queue for an attendant.

- Authorization Codes

The system does not prompt for an authorization code, and the **route-to** command fails if:

- Authorization codes are enabled
- A **route-to** command in a prompting vector accesses either Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS)
- The Facility Restriction Level (FRL) of the vector directory number (VDN) does not have the permission to use the chosen routing preference

- CallVisor Adjunct-Switch Application Interface (ASAI)

The Call Vectoring feature can collect ASAI-provided digits through the **collect vector** command as dial-ahead digits. The system passes CINFO to CallVisor ASAI.

- Hold

If a call is put on hold during the processing of a collect command, the command restarts at 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 an adjunct might later use 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. 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 suspend vector processing in the same way as transferred calls.

Automatic Callback

Use the Automatic Callback (ACB) feature to allow internal users who place a call to a busy or an unanswered internal telephone to be called back 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 automatically 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 that the system provided on the original call.

When you place a call from an analog telephone, and the line is busy, an announcement prompts you to either:

- Enter the digit 1 to activate Automatic Callback.
- Enter the digit 2 to route the call to a hunt group extension.

Automatic Callback supports the following capabilities:

- Ringback Queing
- Analog Busy Automatic Callback Without Flash

Detailed description of Automatic Callback

With the Automatic Callback feature, when a caller places a call to a busy or an unanswered internal telephone, the system calls the caller back when the called telephone becomes available. Upon hearing a busy signal, a user activates Automatic Callback, and hangs up. The system monitors the called telephone. When the called telephone becomes available to receive a call, the system automatically 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 that the system provided on the original call.

To activate this feature, the user of a single-line telephone presses the Recall button, or flashes the switchhook. The user then dials the Automatic Callback feature access code (FAC). A single-line user can activate Automatic Callback for only one call at a time.

The number of call for which a user of a multiappearance telephone can activate Automatic Callback depends on the number of Automatic Callback buttons assigned to the telephone. After the user places a call to a telephone that is busy or unanswered, the user presses an idle Automatic Callback button, and hangs up.

If the original caller answers an Automatic Callback call, and for some reason the called extension cannot accept the Automatic Callback call, the caller hears a confirmation tone and then silence. The call is still queued.

Users cannot activate Automatic Callback for calls to:

- A telephone that is assigned Termination Restriction
- An extension toward which Automatic Callback is already activated
- A data terminal or a data module

- An attendant console group
- A Terminating Extension Group (TEG)
- An extension for a hunt group, a split, or a skill
- The login ID of an EAS agent
- A Vector Directory Number (VDN) Extension

Ringback Queuing

You can administer your system to call users back if users try to place an outgoing call over a trunk group when all trunks are busy. Ringback Queuing places outgoing calls in an ordered queue (first-in, first-out) when all trunks are busy. The system automatically places the callback call to the telephone when a trunk becomes available, and the user hears a distinctive three-burst signal, which indicates an Automatic Callback call from the system. When the user answers the callback call, the original call automatically continues. Redialing is not required. Ringback queuing is also called Automatic Callback for busy trunks.

If a user with a multiappearance telephone has an idle Automatic Callback button and tries to access an all-trunks-busy trunk group, the call queues automatically. The lamp that is associated with the Automatic Callback button lights, and the user hears a confirmation tone.

Ringback Queuing is automatic for a single-line telephone. After dialing is complete, the user hears a confirmation tone if the queue is available. No action is required. The system queues as many calls as are allowed, based on the value in the **Queue Length** field on the **Trunk Group** screen. The system checks the busy or idle status of the trunk group only once. If all trunks are busy, the call queues, even if a trunk has become available by the time that the caller has finished dialing. In this case, a caller might be called back immediately after the caller receives a confirmation tone and hangs up.

You can specify queuing for any outgoing-only trunk group that is not part of a distributed communications system (DCS), or for the outward direction of a non-DCS two-way trunk group.

Analog Busy Automatic Callback Without Flash

With Analog Busy Automatic Callback without Flash, callers who place a call from an analog station to a station that is busy and has no coverage path or forwarding, hear an announcement and are presented with options. Depending on the selection that the caller makes, the call is queued to Automatic Callback, routed to an extension, or dropped. No switchhook flash is required for Analog Busy Automatic Callback.

Hardware requirements for Automatic Callback

The Automatic Callback feature is 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 Communication Manager on an Avaya DEFINITY[®] Server CSI
- Avaya DEFINITY Server SI with an Avaya SCC1 Media Gateway
- Avaya DEFINITY Server CSI with an SCC1 Media Gateway

Administering Automatic Callback

The following steps are part of the administration process for the Automatic Callback feature:

- [Assigning a feature access code \(FAC\) for Automatic Callback](#)
- [Setting the no-answer timeout interval](#)
- [Assigning a feature button for Automatic Callback](#)
- [Setting the queue length for Ringback Queing](#)

This section describes:

- Any prerequisites for administering the Automatic Callback feature
- The screens that you use to administer the Automatic Callback feature
- Complete administration procedures for the Automatic Callback feature

Prerequisites for administering Automatic Callback

You must complete the following actions before you can administer the Automatic Callback feature:

- None

Screens for administering Automatic Callback

Screen name	Purpose	Fields
Feature Access Code (FAC)	Assign a feature access code (FAC) with which to activate or deactivate Automatic Callback.	Automatic Callback Activation/Deactivation
Feature-Related System Parameters	Set the number of times the callback call rings at the calling station before the system cancels the callback call.	Automatic Callback-No Answer Timeout Interval
Station (multiappearance)	Assign a feature button for the Automatic Callback feature.	Buttons/Feature Button Assignments - auto-cback
Trunk Group	Set the queue length for Ringback Queing.	Queue Length

Assigning a feature access code (FAC) for Automatic Callback

To assign a feature access code (FAC) for Automatic Callback:

- 1 Type **change feature-access codes**. Press **Enter**.

The system displays the *Feature Access Codes (FAC)* screen ([Figure 44, Feature Access Code \(FAC\) screen](#), on page 254).

Figure 44: Feature Access Code (FAC) screen

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:	_____
Attendant	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:	_____

- 2 In the **Automatic Callback Activation** field, type the digits of the access code that you want to use to activate the Automatic Callback feature.
- 3 In the **Deactivation** field, type the digits of the access code that you want to use to deactivate the Automatic Callback feature.
- 4 Press **Enter** to save your changes.

Setting the no-answer timeout interval

You can administer the number of times that the callback call rings at the calling station before the callback call is canceled.

To set the no-answer timeout interval:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the *Feature-Related System Parameters* screen ([Figure 45, Feature-Related System Parameters screen](#), on page 255).

Figure 45: Feature-Related System Parameters screen

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/SIP 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?

```

- 2 In the Automatic Callback - No Answer Timeout Interval (rings) field, enter the number of times that you want a callback call to ring at the calling station before the callback call is canceled. In this example the nuber of rings is set to 4.
- 3 Press **Enter** to save your changes.

Assigning a feature button for Automatic Callback

To assign one or more feature buttons for Automatic Callback:

- 1 Type **change station *n***, where ***n*** is the extension of the station for which you want to assign a feature button. If you are adding a new station, type **add station next**.

The system displays the **Station** screen ([Figure 46, Station screen](#), on page 256).

Figure 46: Station screen

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	5:	
2: call-appr	6:	
3: call-appr	7:	
4: auto-cback	8:	

- 2 In the Button Assignments area, enter **auto-cback** in a blank field.
- 3 Repeat Step 2 for as many buttons as you want to assign for Automatic Callback.
- 4 Press **Enter** to save your changes.

Setting the queue length for Ringback Queing

To set the queue length for Ringback Queing:

- 1 Type **change trunk-group n**, where **n** is the number of an existing trunk group for which you want to set the queue length. If you are adding a new trunk group, type **add trunk-group next** .
The system displays the **Trunk Group** screen ([Figure 47, Trunk Group screen](#), on page 257).

Figure 47: Trunk Group screen

add trunk-group next

Page 1 of x

TRUNK GROUP

Group Number: ____

Group Name: _____

Direction: _____

Dial Access? _

Queue Length: ____

Comm Type: _____

BCC: _

TRUNK PARAMETERS

Trunk Type (in/out): _____

Outgoing Dial Type: _____

Digit Treatment: _____

Analog Loss Group: ____

Incoming Dial Tone? _

Incoming Calling Number - Delete: ____

Bit Rate: _____

Disconnect Supervision - In? _ Out? _

Answer Supervision Timeout: ____

Group Type: tie_____

COR: _

Outgoing Display? _

Busy Threshold: ____

Auth Code? _

Trunk Flash? _

ITC? ____

Incoming Rotary Timeout(sec): ____

Incoming Dial Type: _____

Disconnect Timing(msec): ____

Digits: ____

Sig Bit Inversion: none

Digital Loss Group: ____

Insert: _____

Format: _____

Synchronization: ____

Duplex: ____

Receive Answer Supervision? _

CDR Reports: _

TN: ____

TAC: ____

Trunk Signaling Type: ____

Night Service: _____

Incoming Destination: _____

- 2 In the Queue Length field, type the number of outgoing calls that you want to be held waiting when all trunks are busy.
- 3 Press **Enter** to save your changes.

Reports for Automatic Callback

The following reports provide information about the Automatic Callback feature:

- None

Considerations for Automatic Callback

This section provides information about how the Automatic Callback feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Automatic Callback under all conditions. The following considerations apply to Automatic Callback:

- The system cancels an Automatic Callback request if the:
 - Called party is unavailable within 30 minutes.
 - Calling party does not answer the callback call within the administered interval. This interval consists of two to nine ringing cycles, and is set in the **Automatic Callback-No Answer Timeout Interval** field on the **Feature-Related System Parameters** screen.
 - Calling party decides not to wait, and presses the same Automatic Callback button a second time on a multiappearance telephone, or dials the Automatic Callback cancellation code on a single-line telephone.
- Automatic Callback is administered for individual telephones by their Class of Service (COS), and cannot be assigned to attendants. Multiappearance telephones must have an Automatic Callback button to activate the feature.
- Automatic Callback works differently depending on whether the called party is busy or does not answer the call. For a busy call, Automatic Callback occurs as soon as the called party hangs up. For an unanswered call, the telephone must be used for another call, and then hung up before Automatic Callback occurs.

NOTE:

If the user who originates an Automatic Callback originator has all line appearances occupied when the Automatic Callback call comes in, the user hears priority ringing once, and the Automatic Callback lamp blinks. However, if the user presses the Automatic Callback button to answer the Automatic Callback call, the system drops one of the other calls.

- Queuing can reduce the number of trunks required.
- On a multiappearance telephone, one callback call can be associated with each Automatic Callback button assigned to the terminal.
- On a single-line telephone, only one Automatic Callback call can wait at a time.
- Queue requests are canceled when:
 - — A trunk is unavailable within 30 minutes.
 - — The user does not answer the callback call within the administered interval.
 - — The telephone is busy when the callback call is attempted.
 - — The user dials the Ringback Queuing cancellation code, or presses the Automatic Callback button that is associated with the queued call.
- Incoming tie-trunk calls cannot queue on an outgoing trunk group. The system does not know the calling number and cannot originate the callback call.
- The system checks the busy or idle status of the trunk group only once. If all trunks are busy, the call queues, even if a trunk has become available by the time that the caller has finished dialing. In this case, a caller might be called back immediately after the caller receives a confirmation tone and hangs up.
- A trunk might appear to be available, yet outgoing calls are queued. The trunk is not free, because the trunk is reserved for a previous Automatic Callback request.

Interactions for Automatic Callback

This section provides information about how the Automatic Callback feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Automatic Callback in any feature configuration.

- **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.
- **Automatic Route Selection**
If a user with a multiappearance telephone that has an Automatic Callback button makes an ARS call, and all trunks are busy, the system activates Ringback Queuing automatically.
- **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 user has activated Automatic Callback on a telephone, you cannot perform Busy Verification of that telephone.
- **Call Coverage**
The system does not redirect Automatic Callback calls to coverage.
- **Call Forwarding**
If a user has activated Call Forwarding on a telephone, 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 (COR)**
Telephones with origination restriction cannot activate Automatic Callback.
- **Conference and Transfer**
A user of a single-line telephone cannot activate conference or transfer if Automatic Callback is activated.
- **Distributed Communication System (DCS)**
Automatic Callback operates over a DCS network in the same way Automatic Callback operates on a local server.
- **Expert Agent Selection (EAS)**
Users cannot activate Automatic Callback to the login ID of an EAS agent. Users can activate Automatic Callback to the telephone where the agent is logged in.

Automatic Callback

Interactions for Automatic Callback

- Hold

A user of a single-line telephone cannot receive Automatic Callback calls if the user has placed a call on hold.
- Hot Line Service

Telephones that are 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.
- Remote Access

Callback calls cannot be made to Remote Access users, because the system does not know the calling number.
- 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 translated phrase for Administrable Language Displays.

Automatic Circuit Assurance

Use the Automatic circuit assurance (ACA) feature to identify possible trunk malfunctions. When you enable ACA, the system measures the holding time of each trunk call. If the measurements show calls with either extremely long or extremely short holding times, Avaya Communication Manager places a referral call to an attendant or a telephone.

Detailed description of Automatic Circuit Assurance

This section provides a detailed description of the Automatic Circuit Assurance feature (ACA).

The system records the holding time from when a trunk is accessed to when the trunk 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 measure all trunks, or only certain trunks.

Avaya Communication Manager 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, the system 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 limit, 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 central office (CO) lines, out-of-service trunks, or trunks that are undergoing maintenance testing.

The referral call

An ACA call includes a display message or a voice-synthesized message that states:

- That this call is an ACA call
- The access code, the trunk group number, and the trunk group member number
- The type of referral (short holding time or long holding time)

If the referral call is answered, this information is displayed and remains displayed until the call is released. If the call is not answered within 3 minutes, the system ends the call. The system places the call again after 1 hour, and continues to place the call hourly until someone answers.

The attendant or the telephone user who receives the referral call can press the aca-halt button to stop further calls, if one is present. This is a toggle button, and turns off the ACA feature until the user presses the button again.

The audit trail

When the system makes a referral call, the system also adds a record to an audit trail. Audit trail records are available on the ACA Measurements Report. Each record contains the:

- Time and the date of referral
- Trunk group number, the trunk access code, and the trunk group member
- Type of referral (short holding time or long holding time)

Hardware requirements for Automatic Circuit Assurance

The Automatic Circuit Assurance (ACA) feature requires the following hardware:

- None

Administering Automatic Circuit Assurance

This section describes the screens required to administer the Automatic Circuit Assurance feature.

Screens for administering Automatic Circuit Assurance

Screen name	Purpose	Fields
<i>Feature-Related System Parameters</i>	Enable the ACA feature.	• Automatic Circuit Assurance Enabled • ACA Referral Calls • ACA Referral Destination • ACA Short Holding Time Originating Extension • ACA Long Holding Time Originating Extension
<i>Trunk Features</i>	Set assignments and thresholds.	• ACA Assignment • Short Holding Threshold • Long Holding Time (hours) • Short Holding Time (seconds)

Screen name	Purpose	Fields
<i>Attendant Console</i>	Set feature button.	Feature Button – aca-halt
<i>Station</i>	Set feature button.	Feature Button – aca-halt

Reports for Automatic Circuit Assurance

The following reports provide information about the Automatic Circuit Assurance (ACA) feature:

- The ACA Measurements Report shows the audit trail for ACA calls.

Considerations for Automatic Circuit Assurance

This section provides information about how the Automatic Circuit Assurance (ACA) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Automatic Circuit Assurance under all conditions. The following considerations apply to Automatic Circuit Assurance:

- None

Interactions for Automatic Circuit Assurance

This section provides information about how the Automatic Circuit Assurance feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Automatic Circuit Assurance in any feature configuration.

- **Administrable Language Displays**
You cannot administer languages for ACA messages.
- **AUDIX**
Do not set the referral-call extension to a telephone that covers to an AUDIX voice messaging system. You can overload AUDIX with the volume of calls, because ACA calls remain active for up to three minutes.
- **Busy Verification**
Once you identify a potentially defective trunk, you can use Busy Verification to check that trunk.
- **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 a local attendant 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 that indicates the failed trunk is lost, even if the referral call is made over a DCS network.

- Night Service

The system does not place referral calls to the attendant if the system is in Night Service mode.

- Visually Impaired Attendant Service (VIAS)

If the attendant presses the Display Status button and an ACA call has not been answered, 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 the switch contains a voice-synthesis board, ACA referral calls are also accompanied by an audible message that identifies 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 Number Identification

Use the Inband Automatic Number Identification (ANI) to interpret calling party information, such as a calling party number or a billing number.

Detailed description of Automatic Number Identification

This section provides a detailed description of the Automatic Number Identification (ANI) feature.

You use inband signaling for information, such as the address digits for the called party, that is delivered over the same trunk circuit that is used for the voice or data connection.

You use out-of-band or ISDN signaling when signaling information passes through a different signaling path than the path that is used for the voice or data connection.

For example, when a call is made from 555-3800 to your display telephone 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 telephone display.

Outgoing Automatic Number Identification

Outgoing ANI applies to outgoing Russian multifrequency (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 by way of multi-frequency signaling) prefixes by COR. This allows a switch to send different ANIs to different central offices (COs).

For a tandem call that uses different types of incoming and outgoing trunks, the server uses:

- The COR -assigned call type of the incoming trunk for Russian or R2-MFC outgoing trunks
- Automatic Route Selection (ARS) call types for MFE outgoing trunks

Hardware requirements for Automatic Number Identification

The Automatic Number Identification feature requires the following hardware:

- None

Administering Automatic Number Identification

The following steps are part of the administration process for the Automatic Number Identification feature:

- None

Reports for Automatic Number Identification

The following reports provide information about the Automatic Number Identification feature:

- None

Considerations for Automatic Number Identification

This section provides information about how the Automatic Number Identification feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Automatic Number Identification under all conditions. The following considerations apply to Automatic Number Identification:

- None

Interactions for Automatic Number Identification

This section provides information about how the Automatic Number Identification feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Automatic Number Identification in any feature configuration.

- Attendant Console
If an attendant extends a call, the attendant's Class of Restriction (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. The ANI for the server 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 the 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. The ANI for the server is used if the originating endpoint is an incoming trunk.

Authorization Codes

Use the Authorization Codes feature to control the calling privileges of system users. Authorization codes extend control of calling privilege and enhance security for remote access callers. To maintain system security, Avaya recommends that you use authorization codes.

You can use authorization codes to:

- Override a facilities restriction level (FRL) that is assigned to an originating station or trunk
- Restrict individual incoming tie trunks and remote access trunks from accessing outgoing trunks
- Track Call Detail Recording calls for cost allocation
- Provide additional security control

Detailed description of Authorization Codes

When you dial an authorization code, the Facilities Restriction Level (FRL) that is assigned to the extension, the attendant console, the incoming trunk group, or the remote access trunk group in use for the call is replaced by the FRL assigned to the authorization code. The FRL that is assigned to the authorization code functions in the same way as the original. However, the new FRL that is assigned to the authorization code might represent greater or lesser calling privileges than the original FRL. Access to any given facility depends on the restrictions that are associated with the FRL of the authorization code.

Example

The following example shows how authorization codes work with FRLs.

A supervisor is at a desk of an employee, the supervisor wants to make a call that is usually not allowed by the FRL that is assigned to the extension of that employee. The supervisor, however, can still make the call if the supervisor dials an authorization code that is assigned an FRL. The FRL must be an FRL for the type of call that is allowed.

Length of authorization codes

For security reasons, authorization codes must consist of 4 to 13 digits. The number of digits in the codes must be a fixed length for a particular server that is running Avaya Communication Manager.

Once established, the number of digits (4 to 13) in the authorization code remains fixed unless all codes are removed and reentered. All authorization codes used in the system must be the same length.

Using authorization codes

You can administer incoming trunk groups within a system 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 system completes the call as dialed. If the user does not dial an authorization code, or dials an incorrect authorization code, the system routes the call to an attendant. The system can also route the call to an intercept tone based on how the system was administered.

Usually, Direct Inward Dialing (DID) trunks do not require authorization codes. However, you can administer DID trunks to require an authorization code, but you must do this carefully. Different types of calls can terminate at different endpoints, and requiring an authorization code can be confuse the user.

You can also administer a Cancellation of Authorization Code Request (CACR) digit. The CACR digit cancels the 10-second interval between dialing. When the user dials the CACR digit, the system immediately routes the call according to system administration. Incoming trunk calls receive intercept tone or go to the attendant. Other calls receive intercept tone unless the user's FRL is high enough to route the call. A CACR digit from an off-premises extension over DID and Tie trunks use DID and Tie trunk intercept treatment. Internal calls receive intercept tone.

You must not program passwords or authorization codes onto auto dial buttons. Because display telephones display the programmed buttons, the potential exists for an unauthorized person to use the autodial buttons. If you must program passwords or authorization codes onto auto dial buttons, use the ~s (suppress) character to prevent displaying the codes.

AAR and ARS calls

Each authorization code is assigned a Class of Restriction (COR) that contains an associated FRL. Within a system, the FRL that is assigned to the point at which the call originates determines the access privileges that are associated with the call. When an AAR/ARS call is dialed, the system allows or denies the call, based on the FRL of the originating station. You use COR to restrict internal or non-AAR or an ARS calls.

You can assign authorization codes to individual users to specify the level of calling privileges. Such codes work regardless of the originating facility. Once an authorization code is required and dialed on an AAR or an ARS call, the FRL assigned to the authorization code replaces the originating FRL. This new FRL controls and defines the privileges of the user.

An AAR call or an ARS call that a system user originates, or routes over an incoming tie trunk can require a dialed authorization code to continue routing.

When you administer authorization codes, ensure that the user does not have to dial the authorization code more than once. For example, if a user makes an AAR call or an ARS call, and the FRL of the user 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 that is 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 system routes the call through another switch, the user might be required to dial an authorization code again. This type of situation can be avoided through careful administration.

An authorization code might be required on some, but not all, trunk groups. In such cases, the system prompts for an authorization code when the originating FRL is not high enough to access the next available trunk group in the routing pattern.

Hardware requirements for Authorization Codes

The Authorization Codes feature requires the following hardware:

- None

Administering Authorization Codes

The following steps are part of the administration process for the Authorization Codes feature:

- [Setting up Authorization Codes](#)
- [Creating Authorization Codes with a specific Class of Restriction \(COR\)](#)

This section describes:

- Any prerequisites for administering the Authorization Codes feature
- The screens that you use to administer the Authorization Codes feature
- Complete administration procedures for the Authorization Codes feature

Prerequisites for administering Authorization Codes

You must complete the following actions before you can administer authorization codes:

- On the **Optional Features** screen, ensure that the Authorization Codes field is set to **y**. If the authorization code field is set to **n**, your system is not enabled for the Authorization Codes feature and you cannot perform the procedure. Contact your Avaya representative for assistance.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

Screens for administering Authorization Codes

Screen name	Purpose	Fields
Optional Features	Ensure that the Authorization Codes feature is enabled on the system.	Authorization Code
Feature-Related System Parameters	Enable the Authorization Codes feature, and set up other fields to administering the feature.	• Authorization Code Enabled • Authorization Code Length • Display Authorization Code

Screen name	Purpose	Fields
<i>Authorization Codes - COR Mapping</i>	Create an Authorization Code that has a specific COR.	• AC • COR

Setting up Authorization Codes

To set up authorization codes:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the *Feature-Related System Parameters* screen ([Figure 48, Feature-Related System Parameters screen](#), on page 272).

Figure 48: Feature-Related System Parameters screen

change system-parameters features	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** to enable the Authorization Codes feature on a system-wide basis.
- 3 In the **Authorization Code Length** field, type the number of digits that you want for the authorization code.

This field 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 that the system allows. In the example shown, the authorization code has 7 digits. Avaya recommends that authorization codes must have at least 4 digits, but no more than 13.
- 4 In the **Authorization Code Cancellation Symbol** field, leave the default of the pound sign (**#**).

Users must dial this symbol 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**.
When you set this field to **n**, the system does not route a call to the attendant if a user does not dial an authorization code within 10 seconds, or a user dials an invalid authorization code.
- 6** In the **Display Authorization Code** field, type **n**.
When you set this field to set to **n**, the authorization code does not display on the phone sets thus maximizing security.
- 7** Press **Enter** to save your changes.
- 8** Type **change authorization-code n**, where **n** is the authorization code. Press **Enter**.
The system displays the **Authorization Code - COR Mapping** screen ([Figure 49, Authorization Code - COR Mapping screen](#), on page 273).

Figure 49: Authorization Code - COR Mapping screen

[illegible]

- 9** In the **AC** field, type the authorization code that your users must dial.
The number of digits in the code that you type must be the same as the number you assigned in the **Feature-Related System Parameters** screen in Step 3. In this example, the authorization code is **4285193**.
- 10** In the **COR** field, type a COR number from 0 through 95.
In this example, the COR number is **1**.
- 11** Press **Enter** to save your changes.

Creating Authorization Codes with a specific Class of Restriction (COR)

You can use authorization codes that allow a callers to override the calling privileges. For example, you can give a supervisor an authorization code so that they can make calls from a telephone that is usually restricted for these calls. Since each authorization code has a COR, the system uses the COR that is assigned to the authorization code, and the FRL assigned to the COR to override the privileges associated with the restricted phone.

Authorization codes do not override dialed strings that are denied. For example, if your Automatic Route Selection (ARS) tables restrict users from placing calls to destinations that are outside of the country, a caller cannot override the restriction with an authorization code.

To create an authorization code with a specific COR:

- 1** Type **change authorization-code *n***, where ***n*** is the authorization code. Press **Enter**.

The system displays the **Authorization Code - COR Mapping** screen ([Figure 50, Authorization Code - COR Mapping screen](#), on page 274).

Figure 50: Authorization Code - COR Mapping screen

[illegible]

- 2 In the AC field, type the code that you want to use.
- 3 In the COR field, type the COR number.
- 4 Press **Enter** to save your changes.

Reports for Authorization Codes

The following reports provide information about the Authorization Codes feature:

- None

Considerations for Authorization Codes

This section provides information about how the Authorization Codes feature behaves in certain circumstances. Use this information to ensure that you receive maximum benefits of the Authorization Codes feature under all conditions:

- From remote locations users usually access authorization from touch-tone telephones. However, users can also do so from rotary telephones at specified authorization-code-forced locations that follow appropriate trunk administration practices. Rotary station users access attendants using Listed Directory Numbers (LDN) or Remote Access Numbers (RAN), 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. Random codes prevent hackers from deciphering the sequential codes.
- If timeout to attendant does not occur or a Cancellation of Authorization Code Request (CACR) digit codes are dialed instead of authorization codes, the system assumes that invalid authorization codes were dialed and the caller hears intercept tones.
- Authorization codes can have an impact calling privileges. Authorization codes can:
 - Change the FRL of an outgoing call when the FRL is not high enough to access preferred that AAR/ARS assigns. An FRL is assigned to a COR that is associated with user authorization codes. No additional COR data is assigned.
 - Overrides COR for remote access calls that are assigned to barrier codes, when required. if an authorization code is required for remote access calls, the user is assigned the COR of the dialed authorization code, with all connected data, such as the FRL. This COR overrides the COR that is assigned to any required barrier code.
- Incoming trunk calls that require authorization codes do not change user privileges.

Interactions for Authorization Codes

This section provides information about how the Authorization Codes feature interacts with other features on the system. Use this information to ensure that you receive maximum benefits of the Authorization Codes feature in any feature configuration:

- AAR/ARS Partitioning

Class of Restriction (COR) assigns partitioned group numbers and authorization codes can change CORs. Therefore, authorization codes can change Partitioned Group Numbers on incoming remote access calls. For originating calls, the COR of the user determines Partitioned Group Numbers.

- Cancellation of Authorization Code Request

If	Then
CACR =1	• Authorization $\frac{1}{4}$ 1
Network = DEFG1, DEFG3 or DEF ECSR5	• CACR can be #
Network - S85s, DIM switch	• CACR = 1 (default)

- COR and Facilities Restriction Level (FRL)

Authorization codes used for AAR or ARS calls override the associated FRL.

Associated Classes of Restriction determine remote-access user privileges.

- Forced Entry of Account Codes and Call Detail Recording

For 94A LSU and 3B2 CDRU 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. Note that 94A LSU and 3B2 CDRU are no longer supported.

Authorization codes are recorded after destination addresses are dialed. Invalidly dialed authorization codes are recorded, and CDR printouts can be used to determine patterns.

Bridged Call Appearance

Use the Bridged Call Appearance feature to give single-line and multiappearance telephones an appearance of another telephone number. With Bridged Call Appearance, the user can originate, answer, and bridge onto calls to or from the telephone number of another user.

The Bridged Call Appearance feature supports the following capabilities:

- Single-line telephone

A single-line telephone can be the primary telephone, meaning that it can have a bridged appearance on another telephone. Or, it can have a bridged appearance of either another single-line telephone or a multiappearance telephone.

- Multiappearance telephone

A multiappearance telephone can be the primary telephone, meaning that it can have a bridged appearance on another telephone. A multiappearance telephone can have a bridged appearance of a single-line telephone or one or more appearances of a multiappearance telephone. A multiappearance telephone can also have bridged appearances of multiple telephone numbers.

Detailed description of Bridged Call Appearance

A telephone's primary number is the extension assigned when the telephone is administered. The primary number appears in the Extension field on page 1 of the Station screen. On a multiappearance telephone, multiple appearances of this primary number can exist.

A Bridged Call Appearance is an appearance of a primary number on a different telephone. In most ways the bridged call appearance acts like the primary number appearance. For example, when a telephone number is called, you can answer that number at the primary telephone and at all bridged appearances of that number. When a call is received, the primary telephone and all bridged appearances alert visually, with audible ringing as an administrable option. Likewise, a call that is placed from a bridged call appearance carries the display information and the Class of Restriction (COR) of the primary number.

You can use a bridged call appearance like a primary call appearance for most features. For example, Conference, Transfer, Hold, Drop, and Priority Calling from a bridged appearance, just as from a primary call appearance.

Sample uses

The following list describes some situations in which you might want to use bridged call 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 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.
- For visitor telephones
An executive can have another telephone in their office for use by visitors. It might be desirable that the visitor can bridge onto a call that is active on the executive's primary extension. A bridged call appearance makes this possible.
- In service environments
It might be necessary that several people be able to handle calls to a particular extension. For example, several users might be required to answer calls to a hot line number in addition to their normal functions. Each user might also be required to bridge onto existing hot line calls. A bridged call appearance provides this capability.
- For a user who frequently uses telephones that are in different locations
A user might not spend all of their working hours in the same place. For this type of user, it is convenient to have their extension bridged at several different telephones.

Administrable buttons and lamps for multiappearance telephones

You can administer the message lamp and some feature buttons to apply to the primary telephone rather than the number of the telephone they reside on.

- You can administer the message lamp to light on the bridged user's telephone when messages are waiting for the primary telephone.
- 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. Thus, a bridged user can activate or deactivate Call Forwarding from the telephone with the bridged call appearance for all primary and bridged appearances of the extension. 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. Thus, a bridged user can activate Send All Calls for the primary extension user.

Hardware requirements for Bridged Call Appearance

The Bridged Call Appearance feature requires the following hardware:

- None

Administering Bridged Call Appearance

The following steps are part of the administration process for the Bridged Call Appearance feature:

- [Creating a bridged call appearance on a single-line telephone](#)
- [Creating a bridged call appearance on a multiappearance telephone](#)

This section describes:

- Any prerequisites for administering the Bridged Call Appearance feature
- The screens that you use to administer the Bridged Call Appearance feature
- Complete administration procedures for the Bridged Call Appearance feature

Prerequisites for administering Bridged Call Appearance

You must complete the following actions before you can administer the Bridged Call Appearance feature:

- If Data Privacy for data or voice calls is administered, you can prohibit bridging onto voice or data calls with Data Privacy. To prohibit bridging onto Data Privacy calls, enter **y** in the **Prohibit Bridging Onto Calls with Data Privacy** field on the **Feature-Related System Parameters** screen.

To view the **Feature-Related System Parameters** screen, type **display system-parameters features**. Press **Enter**.

- For each coverage path that includes a telephone with bridged appearances you must administer whether you want a call to skip a coverage point if that telephone has already alerted as a bridged appearance. Enter **n** to skip the coverage point in the **Terminate to Coverage Pts. with Bridged Appearance?** field on the **Coverage Path** screen. Enter **y** to alert both a bridged call and a redirected call.

To view the **Coverage Path** screen, type **change coverage path n**, where **n** is the number of the coverage path you want to change. Press **Enter**.

Screens for administering Bridged Call Appearance

Screen name	Purpose	Fields
Feature-Related System Parameters	Ensure data privacy.	Prohibit Bridging Onto Calls with Data Privacy
Coverage Path	Skip a coverage point.	Terminate to Coverage Pts. with Bridged Appearance

Screen name	Purpose	Fields
Station	Create a bridged call appearance on a single-line telephone	• Bridged Call Alerting • Line Appearance • Btn • Ext
	Create a bridged call appearance on a multiappearance telephone	• Per Button Ring Control • Bridged Call Alerting • Auto Select Any Idle Appearance • Button Assignments • Btn • Ext • Ring

Creating a bridged call appearance on a single-line telephone

To create a bridged call appearance on a single-line telephone:

- 1 Type **change station *n***, where *n* is the extension of the single-line telephone on which you want to create the bridged call appearance. Press **Enter**.

The system displays the **Station** screen for a single-line telephone.

- 2 Click **Next** until you see the Bridged Call Alerting field ([Figure 51, Station screen \(single-line\)](#), on page 280).

Figure 51: Station screen (single-line)

STATION		
SITE DATA		
Room: _____	Headset? ☐	
Jack: _____	Speaker? ☐	
Cable: _____	Mounting? ☐	
Floor: _____	Cord Length:	
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:

- 3 Perform one of the following actions:
 - If you want the bridged appearance to ring when a call arrives, enter **y**.
 - If you want the call to flash at the bridged appearance, but not ring, enter **n**.
- 4 Move to the **Line Appearance** field, enter a brdg-appr to create a bridged appearance of a single-line telephone or brdg-appr to create a bridged appearance of a multiappearance telephone. Press **Enter**.
- 5 The system displays the **Btn** and **Ext** fields. In the **Btn** field, enter the button number from the primary telephone that you want to use for the bridged call appearance.
- 6 In the **Ext** field, enter the extension of the primary telephone.
- 7 Press **Enter** to save your changes.

Creating a bridged call appearance on a multiappearance telephone

To create a bridged call appearance on a multiappearance telephone:

- 1 Type **change station n**, where **n** is the extension of the multiappearance telephone on which you want to create the bridged call appearance. Press **Enter**.
The system displays the **Station** screen for a multiappearance telephone.
- 2 Click **Next** until you see the **Per Button Ring Control?** field ([Figure 52, Station screen \(multiappearance telephone\)](#), on page 281).

Figure 52: Station screen (multiappearance telephone)

FEATURE OPTIONS	STATION
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? _	Restrict Last Appearance? _
Active Station Ringing: _____	Data Module? _
XID? _	Fixed TEI? _ TEI: _
MIM Support? _	Endpt Init? _ SPID: _ MIM Mtce/Mgt? _
AUDIX Name: _____	Audible Message Waiting? _
Messaging Server Name: _____	Disp Client Redir? _
	Select Last Used Appearance? _

- 3 Perform one of the following actions:
 - To assign ringing separately to each bridged appearance, type **y**.
 - To assign all bridged appearances to either ring or not ring as determined by the **Bridged Call Alerting** field, leave the default set to **n**.

- 4 In the **Bridged Call Alerting?** field., perform one of the following actions:
 - If you want the bridged appearance to ring when a call arrives at the primary telephone, enter **y**.
 - Leave the default set to **n** if you want the call to flash at the bridged appearance, but not ring.
- 5 In the **Auto Select Any Idle Appearance?** field, perform one of the following actions:
 - Enter **y** to allow automatic selection of an alternate idle appearance for transferred or conferenced calls when another appearance of the bridged number is unavailable.
 - Leave the default set to **n** to prevent this automatic selection.
- 6 Move to the **Button Assignments** field. Next to any available button number, perform the following actions:
 - Enter **abrdg-appr** to create a bridged appearance of a single-line telephone.
 - Enter **brdg-appr** to create a bridged appearance of a multiappearance telephone.
- 7 Press **Enter**. The system displays the **Btn** and **Ext** fields. If **Per Button Ring Control** is set to **y**, the system also displays the **Ring** field ([Figure 53, Station screen \(multiappearance set\)](#), on page 282).

Figure 53: Station screen (multiappearance set)

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:
2:	brdg-appr	Btn:	Ext: Ring:

- 8 In the **Btn** field, enter the button number from the primary phone that you want to use for the bridged call appearance.
- 9 Enter the extension of the primary telephone.
- 10 In the **Ring** field, perform one of the following actions:
 - To assign the bridged appearance to ring when a call arrives at the primary telephone, enter **y**.
 - To assign the bridged appearance to not ring, leave the default set to **n**.
- 11 Press **Enter** to save your changes.

Reports for Bridged Call Appearance

The following reports provide information about the Bridged Call Appearance feature:

- None

Considerations for Bridged Call Appearance

This section provides information about how the Bridged Call Appearance feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Bridged Call Appearance feature under all conditions.

The following considerations apply to single-line telephones:

- A bridging user cannot have more than one bridged appearance for a particular primary telephone. However, a multiappearance bridging user can have bridged appearances of more than one single-line telephone on their telephone (a multiappearance bridging user, by use of different buttons, can bridge onto several different primary telephones).
- 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.
- 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.
- 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 other than the primary extension (for example, terminating extension group (TEG), UCD group, call coverage answer group, or DDC group extension), a bridged call appearance is not maintained. Therefore, the primary telephone should not be made a member of such a group.
- 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 following considerations apply to multiappearance telephones:

- 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. The number of bridged call appearances allowed at each telephone is limited only by the number of 2-lamp buttons available on the telephone.
- Up to six parties can be off-hook and involved in a conversation on a bridged appearance of an extension.
- A bridging telephone might have a bridged call appearance corresponding to each call appearance of the primary extension 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. You can use a Call Coverage module or expansion module to provide additional bridged call appearances.

- If a call terminates at a telephone on an extension other than the primary extension, a bridged call appearance is not maintained. Examples of such termination points can be TEG, UCD group, call coverage answer group, or DDC group extensions. Therefore, the primary telephone should not be made a member of such a group.
- You can administer conference tone, which, when enabled, is heard when two parties are bridged together on an active call with a third party.
- 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.

Interactions for Bridged Call Appearance

This section provides information about how the Bridged Call Appearance feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Bridged Call Appearance in any feature configuration.

- **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 feature access code (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 or from a bridged call appearance of the primary extension is displayed as a call from the primary extension. That is, the call is displayed as coming from the primary extension regardless of which appearance placed the call.
On multiappearance telephones, the display at the primary telephone shows the same information for a bridged call appearance as the information for a nonbridged call. For calls to the primary extension, the display at a zero call appearance bridging telephone shows a call from the originator to the primary with no “redirection reason” character.
- **Automatic Call Distribution (ACD)**
Bridged appearances cannot be accessed using 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 primary telephone. The callback call rings at all bridged appearances of the extension and at the primary telephone. This ring is set with priority call distinctive ringing signal. It displays at all telephones and shows that it is a callback call.
- **Busy Indicator (multiappearance telephones only)**
A user presses a Busy Indicator button to call the extension associated with the Busy Indicator button. When a user presses a Busy Indicator button on a zero primary call appearance telephone, the system uses the first available bridged call appearance to place the call.

- **Call Coverage**
 - **Single-line telephones**

When a single-line 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 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. Such a call does not follow 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 primary telephone from the bridged appearance.

The primary telephone should not be a member of a call coverage group. This is because calls to the primary telephone as a member of the group are not bridged.

You can administer the system so 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 satisfied.

If the primary telephone is a single-line telephone with a bridged call appearance on a multiappearance telephone, an incoming call to the single-line 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. This is because there are no primary call appearances. Therefore, the call redirects to the next available coverage point.
- **Call Detail Recording (CDR)**

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 like 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 CDR, the extension 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 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 from any bridged call appearance of that number. When activated, calls to the primary extension 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.

- **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 pickup 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. This can be done instead of selecting the bridged appearance button. This pickup 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 users 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 pickup 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 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 pickup causes the call to terminate on the bridging user's primary extension button. Calls on the primary telephone and all bridged appearances 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 even if 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.

You can place a call on hold using normal single-line conference procedures when the primary telephone user is active on a call, and no other bridging user is active on the call. Any attempt by a bridging user to bridge onto the call during a successful conference attempt is denied.

A single bridging user can conference the call using the normal multiappearance telephone conference procedures. Any attempt by the single-line 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. The user can also ask 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 conference control 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

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 a primary appearance at that same telephone, the bridged extension becomes disassociated from the conference call. In this case, the primary user 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 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

Calls to the primary telephone as a member of a Coverage Answer 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.
 - 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 a primary 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.

— 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 primary's appearance button and the bridged appearance button show 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 place 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, Avaya does not recommend such assignment because DDC/UCD calls do not terminate at any bridged appearances of that extension on other telephones.

- Intercom (multiappearance telephones only)

Bridged appearances of a primary extension 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 someone is screening all calls for the primary telephone, and is indicating who is calling via intercom, the primary telephone must have a call appearance on which to receive and send intercom calls.

- Internal Automatic Answer (IAA)

Calls terminating to a bridged appearance of an IAA-eligible telephone are not eligible for IAA.

- Last Number Dialed (LND)

Activation of the LND feature causes the last number dialed from the activating telephone to be redialed, regardless of the extension used (primary or bridged call appearance).

- Leave Word Calling (LWC)

A LWC message left by a user on a bridged call appearance leaves a message for the called party to call the primary extension 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.

- 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 on a PCOL call, bridged call appearances of that extension 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.

- **Personal Status Access (PSA)**

PSA allows a user to execute a dissociate request from a bridged appearance. However, when a user executes a dissociate command at telephone B, the user dissociates from telephone B. This is the case, even if the user is on a bridged appearance that belongs to telephone A.

- **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.

- **Privacy-Manual Exclusion**

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 bridged appearances 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 primary user of the bridged extension has activated Ringback Queueing, the resulting callback call alerts at bridged appearances as well as at the primary 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 appearances' extensions.

- **Service Observing**

The primary 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 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 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)**

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. The primary telephone should not be assigned to a TEG.

- **Terminal Translation Initialization (TTI)**

If a user is on a bridged appearance, the user cannot use TTI to separate from the telephone.

- Transfer
 - Single-line telephones

A call cannot be transferred by a single-line 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 single-line 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 a single-line 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.

Bulletin Board

Use the Bulletin Board feature to post information and retrieve messages from other users on the server. Avaya personnel can leave high-priority messages on the bulletin board. The system displays the high-priority messages as the first ten lines on the bulletin board.

Detailed description of Bulletin Board

This section provides a detailed description of the Bulletin Board feature.

Use the Bulletin Board feature to post information and retrieve messages from other users on the server. Anyone with an init or an inad login can add or change messages on the bulletin board. Avaya personnel can also leave high-priority messages about escalations on the bulletin board. The system displays the high-priority messages as the first 10 lines on the bulletin board.

When you log in to the system, the system alerts you to any messages on the bulletin board, and gives you the date of the last message.

If an Avaya employee enters a high-priority message while you are logged in, you receive the notification the next time that you enter a command. This high-priority message disappears after you enter a command. The system displays the high-priority message again each time that you log in, until you remove the message.

The bulletin board provides three pages of message space. You can write on any available line other than the high-priority lines.

It is your responsibility to maintain the bulletin board. If your bulletin board exceeds 80% capacity, the system displays the capacity that remains when you log in. If the bulletin board is full, the new messages overwrite the old messages.

Hardware requirements for Bulletin Board

The Bulletin Board feature requires the following hardware:

- None

Administering Bulletin Board

The following steps are part of the administration process for the Bulletin Board feature:

- [Changing bulletin board information](#)

This section describes:

- Any prerequisites for administering the Bulletin Board feature
- The screens that you use to administer the Bulletin Board feature
- Complete administration procedures for the Bulletin Board feature

Prerequisites for administering Bulletin Board

You must complete the following actions before someone can change bulletin board information:

- Give a user permission to type nonpriority information on the **Bulletin Board** screen.

To give a user permission to type nonpriority information on the **Bulletin Board** screen:

- 1 Type **change permissions login ID**, where **login ID** identifies the user who is to use the Bulletin Board feature. Press **Enter**.

The system displays the **Command Permission Categories** screen ([Figure 54, Command Permission Categories screen](#), on page 294).

Figure 54: Command Permission Categories screen

Page 1 of 3

COMMAND PERMISSION CATEGORIES
Login Name: Sup3ru

COMMON COMMANDS	Display Admin. and Maint. Data? y System Measurements: y	
ADMINISTRATION COMMANDS	Administer Stations? y Administer Trunks? y Additional Restrictions? n	Administer Features? y Administer Permissions? y
MAINTENANCE COMMANDS	Maintain Stations? n Maintain Trunks? n Maintain Systems? n	Maintain Switch Circuit Packs? n Maintain Process Circuit Packs? n Maintain Enhanced DS1? n

- 2 In the **Display Admin. and Maint. Data?** field, perform one of the following actions:
 - Type **y** if you want the user to:
 - Type nonpriority messages on the bulletin board
 - Use the display, list, monitor, status commands
 - Change his or her password
 - Schedule reports

- 3 In the **Administer Features** field, perform one of the following actions:
 - Type **y** if you want the user to:
 - Type nonpriority messages on the bulletin board
 - Administer the feature-related parameters, such as coverage paths, class of service, class of restriction, system parameters, authorization codes, and security
- 4 Press **Enter** to save your changes.

Changing bulletin board information

To change bulletin board information:

- 1 Type **change bulletin-board**. Press **Enter**.

The system displays the **Bulletin Board** screen ([Figure 55, Bulletin Board screen](#), on page 295).

Figure 55: Bulletin Board screen

change bulletin-board	Page 1 of 3
Message (* indicates high-priority)	Date
*Avaya is in the process of	03/02/2004
*investigating your trunk lockup problem.	03/02/2004
*The Bulletin Board will be updated as	03/02/2004
*we find information.	03/02/2004
* We have identified the problem.	03/02/2004
*The trunk you added does not provide	03/02/2004
*disconnect supervision. However, the	03/02/2004
*trunk group was administered as such.	03/02/2004
*Please call Pat J. for details.	03/02/2004
 We recently added a new trunk group (14)	03/02/2004
and have had many of the members getting	03/02/2004
locked up.	03/02/2004
We see the error - thanks for checking.	03/02/2004

The **date** field is a display-only field that contains the date that someone added or changed the line of information.

- 2 Type the bulletin board information.

Those who have an init or an inads login, including Avaya personnel, can type high-priority information in **Line 1** through **Line 10**. High-Priority messages have an asterisk (*) at the beginning of the line. When someone with an init or an inads login types the high-priority information, the system displays the high-priority message the next time that you log into the system.

Those with an init or an inads login can type any of the following characters, number, and symbols:

Table 1: Bulletin board entries

Character, number, or symbol	Symbol Name
A through Z	
a through z	
Blank	
0 through 9	
!	Exclamation mark
@	At sign
#	Pound sign
\$	Dollar sign
%	Percent sign
^	Circumflex
&	Ampersand
*	Asterisk
_	Underscore
+	Plus sign
-	Minus sign or dash
=	Equal sign
[	Left bracket
]	Right bracket
{	Left brace
}	Right brace
	Bar or pipe
\	Back slash
‘	left single-quotation mark
’	right single-quotation mark or apostrophe
~	Tilde
;	Semi-colon
:	Colon
1 of 2	

Table 1: Bulletin board entries

Character, number, or symbol	Symbol Name
,	Comma
”	Right double-quotation mark
<	Left angle-bracket
>	Right angle-bracket
.	Period
/	Forward slash
?	Question mark
2 of 2	

3 Type the bulletin board information.

Those with permission, as defined on the *Command Permission Categories* screen, can type message information in:

- Line 11 through Line 19 of Page 1
- Line 1 through Line 20 of Page 2
- Line 1 through Line 20 of Page 3

Type any of the characters, numbers, or symbols shown in [Table 1, Bulletin board entries](#), on page 296. Press **Enter** to save your changes.

Screens for administering Bulletin Board

Screen Name	Purpose	Fields
<i>Command Permission Categories</i>	Give permission to type nonpriority information on the <i>Bulletin Board</i> screen.	• Display Admin and Maint Data • Administer Features
<i>Bulletin Board</i>	Add or change priority, or nonpriority, information in the system.	All

Reports for Bulletin Board

The following reports provide information about the Bulletin Board feature:

- None

Considerations for Bulletin Board

This section provides information about how the Bulletin Board feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Bulletin Board under all conditions. The following considerations apply to Bulletin Board:

- Only users with an init or an inads login can add or edit high-priority messages.
- Only one user can change a message at a time.
- The bulletin board does not lose information during a system reset at level 1 or level 2. If you save translations, you can restore the information if a system reset occurs at levels 3, 4, or 5.

Interactions for Bulletin Board

This section provides information about how the Bulletin Board feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of the Bulletin Board feature in any feature configuration.

- None

Busy Indicator

Use the Busy Indicator feature to provide 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 (TEG)
- A hunt group, either direct department calling (DDC) or uniform call distribution (UCD)
- Any loudspeaker paging zone, including all zones

Busy Indicator supports the following capabilities:

- Busy Tone Disconnect

Detailed description of Busy Indicator

This section provides a detailed description of the Busy Indicator feature.

You can assign extension numbers, trunk group access codes, and Loudspeaker Paging access codes to a Busy Indicator button.

The Busy Indicator button provides the attendant or the user with direct access to the extension number, trunk group, or paging zone.

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 Central Office (CO) sends a busy tone for the disconnect message. With Busy Tone Disconnect (BTD), the switch disconnects analog loop-start CO trunks when a busy tone is sent from the CO.

A call that originates from or terminates to a telephone that uses a BTD enabled trunk has a Call Classifier port connected to the trunk. The Call Classifier port connects after 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 only one BTD trunk is on a call when the BTD signal is received, the call is dropped. If the call is a conference call, only the trunk is dropped. The rest of the parties stay connected.

Hardware requirements for Busy Indicator

The Busy Indicator feature requires the following hardware:

- None

Reports for Busy Indicator

The following reports provide information about the Busy Indicator feature:

- None

Considerations for Busy Indicator

This section provides information about how the Busy Indicator feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Busy Indicator under all conditions. The following considerations apply to Busy Indicator:

- None

Interactions for Busy Indicator

This section provides information about how the Busy Indicator feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Busy Indicator in any feature configuration.

- Answer Supervision

If Answer Supervision is enabled, set the Answer Supervision timeout field to **0** (zero).

Busy Verification

Use the Busy Verification feature (Verify button) to allow attendants and specific multiappearance telephone users to make test calls to trunks, telephones, and hunt DDC and 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. Users can also use this feature to quickly identify faulty trunks.

Detailed description of Busy Verification

This section provides a detailed description of the Busy Verification feature.

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 SI or DEFINITY CSI, the member number is a 2-digit number; for DEFINITY 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, 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 2, Verification of a telephone](#), on page 302 describes the process.

Table 2: 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 least one member is logged in.) [Table 3, Verification of an ACD Split, UCD Group, or DDC Group](#), on page 302 describes the process.

Table 3: Verification of an ACD Split, UCD Group, or DDC Group

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 4, Verification of a Trunk](#), on page 303 describes the process.

Table 4: 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.

Hardware requirements for Busy Verification

The Busy Verification feature requires the following hardware:

- None

Administering Busy Verification

The following steps are part of the administration process for the Busy Verification feature:

- [Using Busy Verification](#)

This section describes:

- Any prerequisites for administering the Busy Verification feature
- The screens that you use to administer the Busy Verification feature
- Complete administration procedures for the Busy Verification feature

Prerequisites for administering Busy Verification

You must complete the following actions before you can administer the Busy Verification> feature:

- View the Trunk Group screen, ensure that the Dial Access field is y. If this field is not set to y, contact your Avaya representative before you continue with this procedure.

Screens for administering Busy Verification

Screen name	Purpose	Fields
<i>Station</i>	Set up busy verify extension.	Feature Button Assignments

Using Busy Verification

- 1 Type **change station xxxx**, where **xxxx** is the station to be assigned the busy verify button. Press **Enter**.

For this example, enter extension **1014**.

The system displays the **Station** screen ([Figure 56, Station screen](#), on page 304).

Figure 56: Station screen

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:	<u>call-appr</u>	4:	verify__
2:	<u>call-appr</u>	5:	_____
3:	<u>call-appr</u>		

- 2 In the Feature Button Assignments field, 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.

Reports for Busy Verification

The following reports provide information about the Busy Verification feature:

- None

Considerations for Busy Verification

This section provides information about how the Busy Verification feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Busy Verification under all conditions. The following considerations apply to Busy Verification:

- 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-port). In this case, the system generates reorder tone.

Interactions for Busy Verification

This section provides information about how the Busy Verification feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Busy Verification in any feature configuration.

- **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.

Call-by-Call Service Selection

Use the Call-by-Call Service Selection (CBC) feature to allow a single ISDN-PRI trunk group to carry calls to a variety of services. Call-by-Call Service Selection eliminates the need to dedicate each trunk group to a specific service. With Call-by-Call Service Selection, you can 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, this feature can reduce the total number of trunks that you must use.
- Improved service. Features and services are less likely to be blocked.
- Simplified networking. This feature simplifies network engineering. Instead of a per-service basis, you can analyze trunking needs based on total traffic.
- Tracking. Call-by-Call Service Selection calls are measurable.

Detailed description of Call-by-Call Service Selection

This section provides a detailed description of the Call-by-Call Service Selection feature.

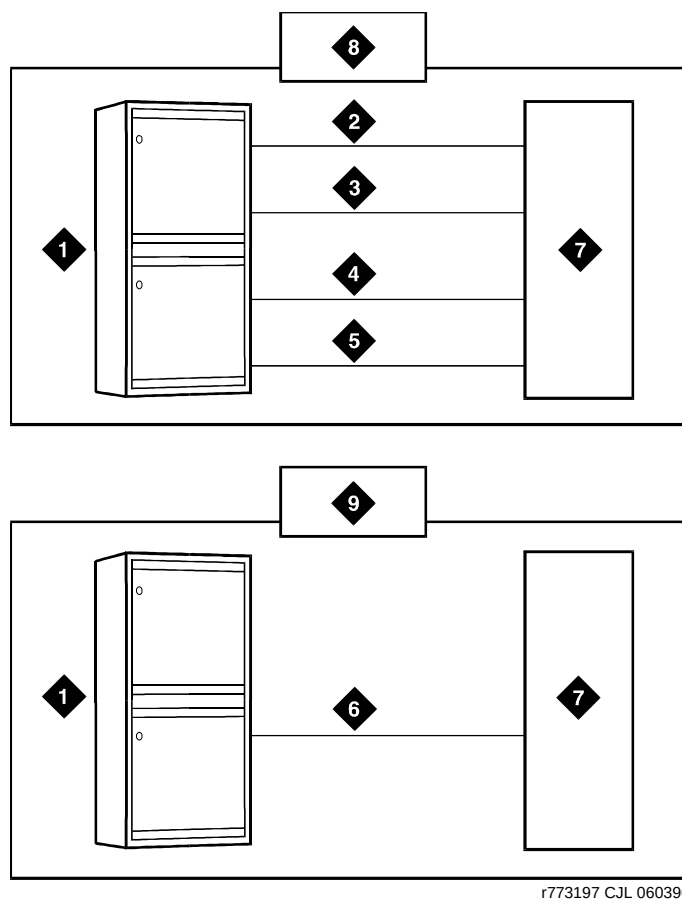
Call-by-Call Service Selection uses the same route patterns and route preferences as Automatic Alternate Routing (AAR), Automatic Route Selection (ARS), and Generalized Route Selection (GRS). The system uses information assigned in the AAR/ARS/GRS route patterns to determine what service or facility to use on an outgoing Call-by-Call Service Selection call.

You can administer a variety of services to use a single trunk group. The system distributes traffic over all available trunks for increased efficiency. Then you can assign services that are used on incoming and outgoing Call-by-Call Service Selection calls.

Using Country Protocol 1, you can integrate services such as MEGACOM, ACCUNET, and INWATS onto a single ISDN-PRI trunk group, with flexible assignment of trunks to each service. Calls such as an incoming 800 service call that requires through-switching as an outgoing WATS call can be routed over the same facility.

[Figure 57, Call-by-Call Service Selection example](#), on page 308 shows an example of Call-by-Call Service Selection.

Figure 57: Call-by-Call Service Selection example



r773197 CJL 060396

Figure notes

1	Avaya Media Server	6	Call-by-Call Service Selection trunk group
2	Megacom trunk group	7	Public switched network
3	Megacom 800 trunk group	8	Without Call-by-Call Service Selection
4	Software-defined network (SDN) trunk group	9	With Call-by-Call Service Selection
5	OUTWATS trunk group		

ISDN messages and information elements for usage allocation

Call-by-Call Service Selection allows the system to specify service types on a call-by-call basis. To specify service types, you assign incoming calls to an ISDN Call-By-Call trunk group based on the number of the called party.

You can also specify service types in a SETUP message. This message indicates that the originating system intends to use the specified service or facility to initiate a call. The SETUP message can contain units called *Information Elements* (IE) that specify call-related information. Call-by-Call Service Selection uses the following IEs:

- Network-Specific Facility (NSF). Indicates which facilities or services are used to complete the call. NSF is usually 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 ensures that the requested service is compatible with the trunk administration before the system accepts the call.

For an outgoing Call-by-Call trunk group, the system uses the service or the feature that is specified on the selected route pattern for the call to construct the NSF IE.

If an associated NSF does not exist for the specified service or feature, the system does not send an NSF IE. For example, SETUP messages for incoming and outgoing calls that are classified only by a called-party number do not contain an NSF IE.

- Transit Network Selection. Indicates which interexchange carrier (IXC) the system uses on an inter-LATA call.

If a call requires both the service or the feature and the IXC to be specified, the system sends the IXC information in the NSF IE, instead of in the Transit Network Selection IE.

Usage Allocation Plans

You can assign Usage Allocation Plans (UAPs) to provide more control over a Call-by-Call Service Selection trunk group. You can allocate a minimum and a maximum number of channels for incoming and outgoing called numbers, privileged users, and voice and data calls.

With a UAP, you can set the:

- Maximum number of trunks that each service can use at any given time. The sum for all services can exceed the total number of trunk group members. For example, for a 15-member trunk group, you can administer a maximum of seven MEGACOM service calls, six MEGACOM 800 service calls, and eight software-defined network (SDN) calls. This ensures that a specific service does not dominate all trunk group members, yet allows for fluctuations in demand.
- Minimum number of trunks that must always be available for each service. The sum for all services cannot 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 service, the minimum number of trunks to use 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 hears a reorder tone, unless other routing preferences are available.

You can assign either a *fixed* or a *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, you can administer different plans to apply at different times of the day and different days of the week. You can assign as many as six activation times and associated plans for each day of the week.

You can administer a simple fixed UAP, or a flexible UAP with many scheduling options. You can even start out with no UAP, and then build the UAP as needed.

Incoming call-handling treatment

Call-by-Call Service Selection provides special incoming call-handling treatment for ISDN trunk groups. The system handles an incoming call on an ISDN trunk according to a treatment table that you administer for the trunk group. Depending on the platform that you use, the table allows for a different number of combinations of call treatments.

The system selects the treatment for an incoming call based on the first three columns of the **Incoming Call-Handling Treatment (ICHT)** page of the **ISDN Trunk Group** screen. When the attributes of an incoming call match these specifications, the system treats the call according to the corresponding entries in the next four columns of the table. 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 $y < x$
Least restrictive	“other”	Specified	Not specified
	“other”	Not specified	Not specified

Call Detail Recording

On successful call attempts that use ISDN Call-By-Call trunk groups, Call Detail Recording (CDR) records the NSF that the NSF IEs of the call specify. CDR refers to this information as the ISDN Network Service (INS). The value that is passed to CDR is the 3-digit equivalent of the NSF IE. The system also records NSF information for Facility Type 2 calls that use ISDN-PRI Call-by-Call trunk groups, 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 (IXC) Code . CDR might not record the IXC properly if the dialed-code format differs from the US IXC code 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 also occurs for the user-defined Facility Type 2. However, the NSF recording occurs only if the NSF is available in the incoming SETUP message.

Hardware requirements for Call-by-Call Service Selection

The Call-by-Call Service Selection feature requires the following hardware:

- ISDN-PRI is supported by the TN767 circuit pack, which is used for assignment of a T1 signaling link and up to 24 ISDN-PRI trunk group members, or by the TN464C circuit pack, which is used for assignment of a T1 or an E1 signaling link. T1 supports up to 24 ISDN-PRI trunk group members, while E1 supports up to 31 members. The TN2207 circuit pack can also be used with ISDN-PRI.
 - For R6si and later configurations, and R6vs and later configurations, the D-channel switches through either the TN765 Processor Interface (PI) circuit pack or the TN778 Packet Control (PACCON) circuit pack.
 - For R6r and later configurations, the D-channel switches through the TN1655 Packet Interface (PKTINT) circuit pack.
 - A TN780 or a TN2182 Tone-Clock circuit pack is required to provide synchronization for the DS1 circuit pack.

NOTE:

You cannot use the TN767 to carry the D-channel if you use either the TN778 (PACCON) or the TN1655 (PKTINT) circuit packs to switch the D-channel. However, in these circumstances, you can use the TN767 for nonfacility -associated signaling (NFAS) interfaces that carry only B-channels.

Administering Call-by-Call Service Selection

You administer Call-by-Call Service Selection on a per-trunk-group basis.

The following steps are part of the administration process for the Call-by-Call Service Selection feature:

- [Setting up a trunk group for CBC](#)
- [Administering route patterns for the CBC trunk group](#)
- [Administering network facilities](#)

This section describes:

- Any prerequisites for administering the Call-by-Call Service Selection feature
- The screens that you use to administer the Call-by-Call Service Selection feature
- Complete administration procedures for the Call-by-Call Service Selection feature

Prerequisites for administering Call-by-Call Service Selection

You must complete the following actions before you can administer the Call-by-Call Service Selection feature:

- View the **Optional Features** screen, and ensure that the following fields are set to **y**:
 - Version
 - ISDN-PRI
 - ISDN-BRI Trunks
 - Usage Allocation Enhancements

If any of these fields is set to **n**, your system might not be enabled for the Call-by-Call Service Selection feature. Contact your Avaya representative before you continue with this procedure.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

Screens for administering Call-by-Call Service Selection

Screen name	Purpose	Fields
Optional Features	Ensure that the proper license fields are set to y .	• Version • ISDN-PRI • ISDN-BRI Trunks • Usage Allocation Enhancements
ISDN Trunk Group	Indicate the service for which this trunk group is dedicated.	• Service Type • Usage Alloc • all fields on the CBC Trunk Group Usage Allocation page • all fields on the CBC Trunk Group Usage Allocation Plan Assignment Schedule page • all fields on the Incoming Call Handling Treatment (ICHT) Table page

Screen name	Purpose	Fields
Route Pattern	Administer the route patterns to use with this trunk group.	- IXC - Service/Feature - Band
Network-Facilities	Administer network facilities for Call-by-call Service Selection.	All

Setting up a trunk group for CBC

To set up a trunk group for Call-by-Call Service Selection:

- 1 Type **change trunk-group *n***, where *n* is the ISDN trunk group that you want to designate for Call-by-Call Service Selection. If you want to add a new trunk group, type **add trunk-group next**.
The system displays the **ISDN Trunk Group** screen ([Figure 58, ISDN Trunk Group screen](#), on page 313).

Figure 58: ISDN Trunk Group screen

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: isdn_____	Auth Code? n	TestCall ITC: rest
Usage Alloc: cbc_____	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: __	
Incoming Calling Number - Delete: __	Insert: _____	Format: _____
Bit Rate: 1200_	Synchronization: async	Duplex: full
Disconnect Supervision - In? y Out? n		
Answer Supervision Timeout: 0_		

- 2 In the **Group Type** field, type **isdn**. The system displays additional ISDN-specific fields.
- 3 In the **Service Type** field, type **cbc** for Call-by-call Service Selection.
- 4 In the **Usage Allocation** field, type **y** to allocate the service that the trunk group provides.
- 5 Press **Next Page** until the system displays the **Incoming Call Handling Treatment** page ([Figure 60, CBC Trunk Group Usage Allocation screen](#), on page 314).

Figure 59: Incoming Call Handling Treatment screen

[illegible]

- 6 Complete the fields on the ***Incoming Call Handling Treatment*** page for each Call-by-Call Service Selection service.
- 7 Press **Next Page** until the system displays the ***CBC Trunk Group Usage Allocation*** page (Figure 60, [CBC Trunk Group Usage Allocation screen](#), on page 314).

Figure 60: CBC Trunk Group Usage Allocation screen

[illegible]

- 8 In the **Service/Feature** field under the **Usage Allocation Plan 1** column, type the name of the service for which you want to allocate CBC trunk service. You can administer up to three Usage Allocation Plans (UAPs) for each trunk.

- 9 In the Min # Chan and Max # Chan fields, type a minimum number and a maximum number of channels for incoming and outgoing called numbers, privileged users, and voice and data calls.
- 10 Repeat Steps 6 and 7 for each service for which you want to allocate CBC trunk service.
- 11 Press **Next Page** until the system displays the **CBC Trunk Group Usage Allocation Plan Assignment Schedule** page ([Figure 60, CBC Trunk Group Usage Allocation screen](#), on page 314).

Figure 61: CBC Trunk Group Usage Allocation Plan Assignment Schedule screen

add trunk-group next
Page y of x

CBC SERVICE TRUNK GROUP ALLOCATION PLAN ASSIGNMENT SCHEDULE

Usage Method:

Fixed? y
 Scheduled? n

Allocation Plan Number: 1

Usage Allocation Plan Activation Schedule:

	Act Time	Plan #	Act Time	Plan #	Act Time	Plan #	Act Time	Plan #	Act Time	Plan #
Sun	—:—	—	—:—	—	—:—	—	—:—	—	—:—	—
Mon	—:—	—	—:—	—	—:—	—	—:—	—	—:—	—
Tue	—:—	—	—:—	—	—:—	—	—:—	—	—:—	—
Wed	—:—	—	—:—	—	—:—	—	—:—	—	—:—	—
Thu	—:—	—	—:—	—	—:—	—	—:—	—	—:—	—
Fri	—:—	—	—:—	—	—:—	—	—:—	—	—:—	—
Sat	—:—	—	—:—	—	—:—	—	—:—	—	—:—	—

- 12 In the Fixed field, perform one of the following actions:
 - If you want a specific UAP to be activated at all times for this trunk group, type **y**. If you type **y**, the Allocation Plan Number field appears.
 - If you do not want a fixed UAP to be activated at all times for this trunk group, type **n**.
- 13 In the Allocation Plan Number field, type the number of the UAP that you want to be activated at all times for this trunk group.
- 14 In the Scheduled field, type **y** if you want to administer a schedule that can change up to six times a day for each day of the week.
- 15 For each day of the week, use the Act Time and the Plan # fields to type the activation time, and type the UAP to use at different times of the day.
- 16 Press **Enter** to save your changes.

Administering route patterns for the CBC trunk group

To administer route patterns for a CBC trunk group:

- 1 Type **change route-pattern n**, where **n** is the number of the route pattern that you want to administer.
The system displays the **Route Pattern** screen ([Figure 62, Route Pattern screen](#), on page 316).

Figure 62: Route Pattern screen

```

change route-pattern 1
                                     Pattern Number: 1_

Grp.  FRL  NPA  Pfx  Hop  Toll  No.  Del  Inserted  DCS/  IXC
No.    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

```

- 2 In the **IXC** (Interexchange Carrier) field, identify the carrier, such as AT&T, that the system uses for calls that are routed over an IXC, and for Call Detail Recording (CDR).
- 3 In the **Service/Feature** fields, type the name of the service that is associated with this route pattern.
- 4 If the value in the **Service/Feature** field is **outwats-bnd**, use the **Band** field to enter a number that represents the OUTWATS band number (US only).
- 5 Press **Enter** to save your changes.

NOTE:

For more information on the **Route Pattern** screen, see the Screen Reference section of the *Administrator's Guide for Avaya Communication Manager*.

Administering network facilities

The **Network Facilities** screen supports the Call-by-Call Service Selection feature for ISDN trunks. Only Avaya personnel can administer these predefined services and features. If the Usage Allocation Enhancement field on the **Optional Features** screen is set to y, you can administer the Additional Services/Features fields.

To administer network facilities:

- 1** Type change isdn network-facilities.

The system displays the **Network Facilities** screen ([Figure 63, Network Facilities screen](#), on page 317).

Figure 63: Network Facilities screen

change isdn network-facilities			Page 1 of X		
NETWORK-FACILITIES					
Facility			Facility		
Name	Type	Coding	Name	Type	Coding
sub-operator	0	00110	mega800	1	00010
operator	0	00101	megacom	1	00011
outwats-bnd	1	00001	inwats	1	00100
sdn	1	00001	wats-max-bnd	1	00101
accunet	1	00110	lds	1	00111
i800	1	01000	multiquest	1	10000
_____	-	_____	_____	-	_____
_____	-	_____	_____	-	_____
_____	-	_____	_____	-	_____
_____	-	_____	_____	-	_____
_____	-	_____	_____	-	_____
_____	-	_____	_____	-	_____
_____	-	_____	_____	-	_____
_____	-	_____	_____	-	_____
_____	-	_____	_____	-	_____
_____	-	_____	_____	-	_____
_____	-	_____	_____	-	_____
_____	-	_____	_____	-	_____
_____	-	_____	_____	-	_____

2 In the **Name** field, type up to 15 alphanumeric characters to specify the name of the indicated service or feature.

3 In the **Facility Type** field, perform one of the following actions:

- If the associated entry is a feature, type **0**.
- If the associated entry is a service, type **1**.
- If the associated entry is of type **incoming**, type **2**.
- If the associated entry is of type **outgoing**, type **3**.

NOTE:

You can administer types 2 and 3 if the **Usage Allocation Enhancements** field on the **Optional Features** screen is set to **y**.

If the **Facility Type** field is set to either **0** or **1**, the **Facility Coding** field displays five binary values. These values specify the ISDN encoding value of the associated service or feature.

4 Press **Enter** to save your changes.

Reports for Call-by-Call Service Selection

The following reports provide information about the Call-by-Call Service Selection feature:

- None

Considerations for Call-by-Call Service Selection

This section provides information about how the Call-by-Call Service Selection feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Call-by-Call Service Selection under all conditions. The following considerations apply to Call-by-Call Service Selection:

- None

Interactions for Call-by-Call Service Selection

This section provides information about how the Call-by-Call Service Selection feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Call-by-Call Service Selection in any feature configuration.

- Call Detail Recording (CDR)

On successful call attempts that use ISDN Call-By-Call trunk groups, CDR records the (Network-Specific Facility) NSF specified by the NSF IE of the call. CDR refers to this information as the ISDN Network Service (INS). The value passed to CDR is the 3-digit equivalent of the NSF IE. NSF information for Facility Type 2 calls, which is used with ISDN-PRI Call-by-Call trunk groups, is also recorded if the NSF is available in the incoming SETUP message.

If an outgoing Call-by-Call Service Selection call uses an interexchange carrier (IXC) other than the presubscribed common carrier, CDR records the 3-digit or 4-digit Interexchange Carrier (IXC) code. CDR might not record the IXC code properly if the dialed-code format differs from the IXC formats that are used in the US.

When a Call-by-Call Service Selection call is rejected because of a Usage Allocation Plan (UAP), CDR records the cause as an ineffective call attempt. The NSF recording also occurs for the user-defined Facility Type 2 call. However, the NSF recording occurs only if the NSF is available in the incoming SETUP message.

- Generalized Route Selection (GRS)

Call-by-Call Service Selection uses the same routing tables and routing preferences that GRS uses.

- Multiquest Flexible Billing

Do not use a service or a facility with the Facility Type 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 the day of week.

- Traffic Measurements

The system provides traffic measurements for each service that is administered for an ISDN Call-by-Call Service Selection trunk group.

Call Charge Information

Use the Call Charge Information feature to determine the approximate charge for calls that are made on outgoing trunks.

Call Charge Information supports the following capabilities:

- **Advice of Charge (AOC)**
AOC collects charge information from the public network for each outgoing call on ISDN trunks.
- **Periodic Pulse Metering (PPM)**
PPM accumulates pulses that are transmitted from the public network at periodic intervals during an outgoing call on a non-ISDN trunk.
- **Charge Display**
Users can display the charges for a call during the call, or when the call ends.

Detailed description of Call Charge Information

This section provides a detailed description of the Call Charge Information feature.

Avaya Communication Manager 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 that represents the cost of a call. The system records the charge information as either a charging unit or currency unit.
- **Periodic Pulse Metering, for non-ISDN trunks**
Periodic Pulse Metering (PPM) accumulates pulses that are 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 on which charges are determined.

Call-charge information helps you to account for the cost of outgoing calls before you receive the call charges from your network provider. The ability to account for the cost of outgoing calls before you receive the call charges from your network provider is especially important in countries where telephone bills are not itemized.

You can also use the information about the cost of outgoing calls to give the cost of telephone calls to employees. The cost information of outgoing telephone calls can help users to better manage their use of company telecommunications facilities. Note, however, that you cannot necessarily use the call charge information that the Call Charge Information feature provides to reconcile telephone bills with your network provider.

You must request either AOC service or PPM service from your network provider. In some areas, your selection is limited. Note that public network does not offer AOC service and PPM service in some countries, including the U S. In some countries, AOC information is received automatically for each call.

Call Charge Information

Detailed description of Call Charge Information

In other countries, the system must request AOC information for each call. Your Avaya representative can help you determine the type of service that you need.

In some countries, the public network sends call-charge information only at the end of a call. In other countries, the public network sends information both at the end of the call and while the call is in progress. PPM is available over the following trunk types:

- Central office (CO)
- Direct inward and outward dialing (DIOD)
- Foreign exchange (FX)
- Personal Central Office Line (PCOL)
- Wide Area Telecommunications Service (WATS)

Charge Display

With Avaya Communication Manager, you can view call-charge information on a telephone display or on a Call Detail Recording (CDR) report.

Charge Display at a user telephone

You can allow users to view call charges on telephone displays. From a display, a user 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 users to control when the display of the call charge information, you can assign a display button that the user can press to see the current call charges. You can also administer the system so that the system displays call charges automatically whenever a user places an outgoing call.

Charge Display on a CDR report

You can administer the system to display call charges on CDR reports. For more information, see the “Call Detail Recording” feature.

Either the ISDN C C field or the PPM field in the CDR record contains the last cumulative charge received from the network. If Call Splitting or Attendant Call Recording is enabled, and a call is transferred for the first time, the ISDN Call Charge field contains the cumulative charge that was most recently received from the network.

For all subsequent transfers, the ISDN Call Charge field contains the difference between the cumulative charge that was most recently received and the value that was 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 that was received
- The outgoing trunk group is not administered for AOC or PPM

Hardware requirements for Call Charge Information

The Call Charge Information feature requires the following hardware:

- None

Administering Call Charge Information

The following steps are part of the administration process for the Call Charge Information feature:

- [Administering the charge display](#)
- [Administering AOC for ISDN trunks](#)
- [Administering PPM for non-ISDN trunks](#)
- [Administering PPM for DS1 circuit packs](#)

This section describes:

- Any prerequisites for administering the Call Charge Information feature
- The screens that you use to administer the Call Charge Information feature
- Complete administration procedures for the Call Charge Information feature

Prerequisites for administering Call Charge Information

You must complete the following actions before you can administer the Call Charge Information feature:

- Defining Call Detail Recording (CDR) to support call charge Information
- Specifying the frequency of the call charge displays
- Translating the text for “Call Charge” for a language other than English
- Assign a Class of Restriction (COR) for automatic charge displays

Defining CDR to support Call Charge information

To define Call Detail Recording (CDR) to support call charge Information:

- 1 Type **change system-parameters cdr**. Press **Enter**.

The system displays the **CDR System Parameters** screen) [Figure 64](#)).

Figure 64: CDR System Parameters screen

change system-parameters cdr	Page 1 of 2
CDR SYSTEM PARAMETERS	
Node Number (Local PBX ID): 18	CDR Date Format: month/day
Primary Output Format: customized	Primary Output Endpoint: CDR1
Secondary Output Format:	
Use ISDN Layouts? n	
Use Enhanced Formats? n	Condition Code 'T' For Redirected Calls? n
Modified Circuit ID Display? n	Remove # From Called Number? y
Record Outgoing Calls Only? n	Intra-switch CDR? y
Suppress CDR for Ineffective Call Attempts? y	Outg Trk Call Splitting? y
Disconnect Information in Place of FRL? n	Outg Attd Call Record? y
	Interworking Feat-flag? y
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 Agent ID on Incoming? n	Record Agent ID on Outgoing? y
Inc Trk Call Splitting? y	Inc Attd Call Record? n
Record Non-Call-Assoc TSC? n	Call Record Handling Option: warning
Record Call-Assoc TSC? n	Digits to Record for Outgoing Calls: outpulsed
Privacy - Digits to Hide: 0	CDR Account Code Length: 4

- 2 Administer the CDR System Parameters screen. For more information, see the “Call Detail Recording” feature.

Specifying the frequency of the call charge displays

To specify the frequency of the charge display:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the *Feature-Related System Parameters* screen ([Figure 65](#)).

Figure 65: Feature-Related System Parameters

display system-parameters features	Page 9 of 14
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? n
Allow Conference via Flash? y	
Vector Disconnect Timer (min):	Network Feedback During Tone Detection? y
	System Updates Time On Station Displays? y
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	
Onhook Dialing on 607/2420/4600/6400/8400 Terminals? n	
ITALIAN DCS PROTOCOL	
Italian Protocol Enabled? n	

- 2 In the **Charge Display Update Frequency (seconds)** field, type the number of seconds, between the charge update information displays that a user sees.

The valid entries are **10** through **60**, and blank.

Note that frequent display updates can have one impact the performance of the system. If the duration of a call is less than the value in the **Charge Display Update Frequency (seconds)** field, the display does not automatically show the charge information.

If you want a user to see the charge information, you must assign a disp-chrg feature button on the **Station** screen for a user.

- 3 Press **Enter** to save your changes.

Translating the text “Call Charge”

To translate the text for “Call Charge:”

- 1 Type **change display-messages miscellaneous-features**. Press **Enter**.

The system displays the *Language Translations* screen ([Figure 66](#)).

Figure 66: Language Translations screen

change display-messages miscellaneous-features		Page 5 of 8
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: *****	

- 2 Page through the screens until you see translation number 47.
- 3 In the Translation field, type the translation for “Call Charge” if you need a translation for the text.
- 4 Press **Enter** to save your changes.

Assigning a COR for charge displays

To Assign a Class of Restriction (COR) for automatic charge display:

- 1 Type **change cor *n***, where *n* is the number of the COR to which you want to assign an automatic charge display. Press **Enter**.

The system displays the *Class of Restriction* screen ([Figure 67](#)).

Figure 67: Class of Restriction screen

display cor 1	Page 1 of 4
CLASS OF RESTRICTION	
COR Number: 1	
COR Description:	
FRL: 7	APLT? y
Can Be Service Observed? n	Calling Party Restriction: none
Can Be A Service Observer? n	Called Party Restriction: none
Time of Day Chart: 2	Forced Entry of Account Codes? n
Priority Queuing? n	Direct Agent Calling? n
Restriction Override: none	Facility Access Trunk Test? n
Restricted Call List? y	Can Change Coverage? n
Access to MCT? y	Fully Restricted Service? n
Group II Category For MFC: 7	Add/Remove Agent Skills? n
Send ANI for MFE? n	Automatic Charge Display? y
MF ANI Prefix:	PASTE (Display PBX Data on Phone)? n
Hear System Music on Hold? y	Can Be Picked Up By Directed Call Pickup? n
	Can Use Directed Call Pickup? n
	Group Controlled Restriction: inactive

- 2 In the Automatic Charge Display field, perform one of the following actions:
 - If you want the user to automatically see the call charges during a call and at the end of a call, type **y**.
 - If you want the user to press the disp-chrg feature button to see the call charges. To see the call charges, the user must press the disp-chrg feature button before the system drops the call, type **n**.
- 3 Press **Enter** to save your changes.

Screens for administering Call Charge Information

Screen name	Purpose	Fields
<i>Attendant Console</i>	Assign a feature button for the attendant to display charges.	FEATURE BUTTON ASSIGNMENTS • disp-chgs
<i>CDR System Parameters</i>	Define CDR for the system.	All
<i>Class of Restriction</i>	Specify a Class of Restriction (COR) that allows the users to have charges displayed automatically.	Automatic Charge Display
<i>DS1 Circuit Pack</i>	Specify the values and the increments for Periodic Pulse Metering (PPM).	• Received Digital Metering Pulse Maximum • Received Digital Metering Pulse Minimum • Received Digital Metering Pulse value
<i>Feature-Related System Parameters</i>	Specify the frequency of the system displays for charge information.	Charge Display Update Frequency
<i>Language Translations</i>	Specify a translation of the “Call Charge” text display.	Call Charge
<i>Station</i>	Assign a feature button for the user to display charges.	Feature Button • disp-chrg
	Assign a COR for an automatic display of call charges.	• COR
<i>Trunk Group (all types)</i>	Specify charge information that controls Call Charge Information processes and displays.	• Charge Conversion • Charge Type • Currency Symbol • Decimal Point

Screen name	Purpose	Fields
Trunk Group • <i>central office (CO)</i> • <i>Direct inward and outward dialing (DIOD)</i> • <i>foreign exchange (FX)</i> • <i>Personal Central Office Line (PCOL)</i> • <i>Wide Area Telecommunications Service (WATS)</i>	Administer PPM for a non-ISDN network.	• CDR Reports • Direction • Frequency • Outgoing Glare Guard (msec) • PPM
Trunk Group-ISDN-PRI	Administer Advice of Charge (AOC) for an ISDN network.	• Charge Advice • Service Type • Supplementary Service Protocol • CDR Reports

Administering the charge display

To administer the charge display, you must complete the following procedures:

- [Administering a trunk group for call charge displays](#)
- [Assigning a call charge display feature button for an attendant](#)
- [Assigning a call charge display feature button for an attendant](#)

Administering a trunk group for call charge displays

To administer a trunk group for call charge displays:

- 1 Type **change trunk-group *n***, where *n* is the number of the trunk-group for which you want to administer call charge display information. Press **Enter**.

The system displays the **Trunk Group** screen ([Figure 68](#)).

Figure 68: Trunk Group screen

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: ____		

- 2 Page through the screens until you see the **Charge Conversion** field.
- 3 In the **Charge Conversion** field, type the charge unit for the currency that you use.

Valid entries are **1 to 64,500**.

The software multiplies the number of charge units by the value of the **Charge Conversion** field, and displays the result as a currency. If the **Charge Conversion** field is blank, the software displays the number of charge units, but does not convert the charge units to a currency.

The system displays the **Charge Conversion** field for central office (CO), direct inward and outward dialing (DIOD), foreign exchange (FX), and Wide Area Telecommunications Service (WATS) trunk groups when the **Direction** field on the **Trunk Group** screen is set to **outgoing** or **two-way**.

The system displays the **Charge Conversion** field for ISDN trunk groups when the **Charge Advice** field on the **Trunk Group** screen is not set to **none**.

- 4 In the **Charge Type** field, type the words or the characters that you want the system to display after the system displays the call charge amount.

You can leave the field blank, or you can type one to seven characters. The system counts a leading space, or an embedded space, as a character.

The system displays the **Charge Type** field for CO, DIOD, FX, and WATS trunk groups when the **Direction** field on the **Trunk Group** screen is set to **outgoing** or **two-way**.

The system displays the **Charge Type** field for ISDN trunk groups when the **Charge Advice** field on the **Trunk Group** screen is not set to **none**.
- 5 In the **Currency Symbol** field, type the symbol that you want the system to display before the system displays the call charge amount.

You can leave the field blank or you can type one to seven characters. The system counts a leading space, or an embedded space, as a character.

The system displays the **Currency Symbol** field for CO, DIOD, FX, and WATS trunk groups when the **Direction** field on the **Trunk Group** screen is set to **outgoing** or **two-way**.

The system displays the **Currency Symbol** field for ISDN trunk groups when the **Charge Advice** field on the **Trunk Group** screen is not set to **none**.
- 6 In the **Decimal Point** field, type the representation of a decimal point that is appropriate for your currency.

You can type **comma**, **period**, or **none**.

If you type **comma** or **period**, the system divides the call charge amount by 100.

The system displays the **Decimal Point** field for CO, DIOD, FX, and WATS trunk groups when the **Direction** field on the **Trunk Group** screen is set to **outgoing** or **two-way**.

The system displays the **Decimal Point** field for ISDN trunk groups when the **Charge Advice** field on the **Trunk Group** screen is not **none**.
- 7 Press **Enter** to save your changes.

Assigning a call charge display button for a user

To assign a call charge display button for a user:

- 1 Type **change station *n***, where *n* is the extension of the user to whom you want to assign a call charge display feature button. Press **Enter**.

The system displays the **Station** screen ([Figure 69, Station screen](#), on page 330) and ([Figure 70, Station screen](#), on page 330).

Figure 69: Station screen

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?	
Model:		
Survivable GK Node Name:	Media Complex Ext:	
	IP Softphone? y	
	Remote Office Phone? y	

Figure 70: Station screen

change station 32009		Page 3 of 4
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	7: disp-chrg	
2: call-appr	8:	
3: call-appr	9:	
4:	10:	
5:	11:	
6: headset	12: release	

- 2 In the COR field, type the number of the COR that supports the automatic display of call charges.
- 3 Page through the screens until you see the **BUTTON ASSIGNMENTS** area.
- 4 In the **BUTTON ASSIGNMENTS** area, type **disp-chrg** next to the feature button number that you want the user to use to display a call charge amount.
- 5 Press **Enter** to save your changes.

Assigning a call charge display feature button for an attendant

To assign a call charge display feature button for an attendant:

- 1 Type **change attendant *n***, where *n* is the number of the attendant console to which you want to assign a charge display feature button.

The system displays the Attendant Console screen ([Figure 71](#)).

Figure 71: Attendant Console screen

change attendant 1

Page 3 of 4

FEATURE BUTTON ASSIGNMENTS		ATTENDANT CONSOLE	
1: split_____		13: disp-chrg_	
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__ *	

- 2 Page through the screens until you see the FEATURE BUTTON ASSIGNMENTS area.
- 3 In the FEATURE BUTTON ASSIGNMENTS area, type **disp-chrg** next to the feature button number that you want the attendant to use to display a call charge amount.
- 4 Press **Enter** to save your changes.

Administering AOC for ISDN trunks

To administer Advice of Charge (AOC) for an ISDN trunk:

- 1 Type **add trunk-group next**. Press Enter.

The system displays the *Trunk Group* screen ([Figure 72](#)).

Figure 72: ISDN Trunk Group screen

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: __	
Incoming Calling Number - Delete: __	Insert: _____	Format: _____
Bit Rate: 1200_	Synchronization: async	Duplex: full
Disconnect Supervision - In? y Out? n		
Answer Supervision Timeout: 0_		

- 2 In the Charge Advice field, perform one of the following actions:
 - If your public network sends AOC automatically, type **automatic**.
 - If the system must request charge information for each call, and you want to receive only the final call charge, type **end-on-request**.
 - If the system must request charge information for each call, and you want the system to display call charges both during the call and at the end of the call, type **during-on-request**.

You can change this field from the default value of none, only if the CDR Reports field is set to y.

- 3 In the Service Type field, type **public-ntwrk**.
- 4 In the Supplementary Service Protocol field, type the supplementary service protocol that this trunk uses. [Table 5, Supplementary service protocols](#), on page 333 shows the valid entries for this field. You can type only one entry.

Table 5: Supplementary service protocols

Valid entries	Usage
a	National
b	ISO/ETSI QSIG Private Network. Also used for the separation of bearer and signaling (SBS) signaling trunks.
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	Distributed Communications System (DCS) with Rerouting Do not use this option if the Service Type field is set to dmi-mos or sddn. Set the Used for DCS field to y .
f	ISDN Feature Plus Public network feature plus signaling.
g	American National Standards Institute (ANSI). Available only if the ISDN - PRI field or the ISDN - BRI field on the Optional Features screen is set to y .

- 5 In the **CDR Reports** field, type the entry that provides the CDR information that you want for your system. [Table 6, CDR report entries](#), on page 333 shows the valid entries for this field.

Table 6: CDR report entries

Valid entries	Usage
y	All outgoing calls on this trunk group generate call detail records. If the Record Outgoing Calls Only field on the CDR System Parameters screen is set to n , incoming calls on this trunk group also generate call detail records.
n	Calls over this trunk group do not generate call detail records.
r	The system generates call detail records for both incoming and outgoing calls. The system also generates the following ringing interval call detail records: - Abandoned calls - The system creates a record with condition code <i>H</i>. Condition code <i>H</i> indicates the time until the call was abandoned. - Answered calls - The system creates a record with condition code <i>G</i>. Condition code <i>G</i> indicates the interval between the call starts to ring and the user answers the call. - Calls to busy stations - The system creates a record with condition code <i>I</i>. Condition code <i>I</i> indicates a recorded call interval of zero.

- 6 Press **Enter** to save your changes.

Administering PPM for non-ISDN trunks

To administer PPM for non-ISDN trunks:

- 1 Type **change trunk-group *n***, where *n* is the number of the trunk-group for which you want to administer periodic pulse metering (PPM). Press **Enter**.

The system displays the **Trunk Group** screen ([Figure 68, Trunk Group screen](#), on page 328) and ([Figure 73](#)).

Figure 73: Trunk Group screen

ADMINISTRABLE TIMERS	
Send Incoming/Outgoing Disconnect Timers to TN465 Ports? _	
Incoming Glare Guard(msec): ____	Outgoing Dial 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
OUTPUTSING INFORMATION	
PPS: 10	Make(msec): 40 Break(msec): 60 PPM? y Frequency: 50/12k

- 2 In the **CDR Reports** field, type the entry that specifies the circumstances under which you want the system to generate CDR report information. [Table 6, CDR report entries](#), on page 333 shows the valid entries for this field.

Table 7: CDR report entries

Valid entries	Usage
y	All outgoing calls on this trunk group generate call detail records. If the Record Outgoing Calls Only field on the CDR System Parameters screen is set to n , incoming calls on this trunk group also generate call detail records.
n	Calls over this trunk group do not generate call detail records.

1 of 2

Table 7: CDR report entries

Valid entries	Usage
r	The system generates call detail records for both incoming and outgoing calls. The system also generates the following ringing interval call detail records: - Abandoned calls - The system creates a record with condition code <i>H</i>. Condition code <i>H</i> indicates the time until the call was abandoned. - Answered calls: The system creates a record with condition code of <i>G</i>. Condition code <i>G</i> indicates the interval between the call starting to ring and the user answering the call. - Calls to busy stations: The system creates a record with condition code <i>I</i>. Condition code <i>I</i> indicates a recorded call interval of zero.
2 of 2	

3 Page through the screens until you see the **Direction** field.

4 In the **Direction** field, perform one of the following actions:

- If you want this trunk to be used for incoming traffic, type **incoming**.
- If you want this trunk to be used for outgoing traffic, type **outgoing**.
- If you want this trunk to be used to network call redirection, type **two-way**.

The system displays this field for all trunk groups except direct inward dialing (DID) and customer-premises equipment (CPE).

5 In the **Glare** field, type the minimum acceptable interval, in milliseconds, between the time that the server sends an outgoing seizure request and when the server receives a seizure acknowledgment. If the interval in which the server receives an acknowledgment is less than the interval that you specify in this field, glare is assumed. Valid entries are the numbers **40** to **100**, in increments of 10.

Only TN2140 ports receive this timer.

You can administer this field only if the **Trunk Type** field is set to **cont**, and the **Direction** field is set to **two-way** or **outgoing**.

6 In the **Frequency** field, type the PPM pulse frequency that the public network requires. [Table 8, PPM pulse frequencies](#), on page 335 shows the valid entries for this field.

The system displays this field only if the **Direction** field is set to **outgoing** to **two-way**, and the PPM field is set to **y**.

Table 8: PPM pulse frequencies

Valid entries	Usage
12k	The TN465B, or later, circuit pack and the TN2184 circuit pack can detect only 12k and 16kHz PPM. If you type 12k in the Frequency field, the circuit pack is set to detect 12kHz.
1 of 2	

Table 8: PPM pulse frequencies

Valid entries	Usage
16k	The TN465B, or later, circuit pack and the TN2184 circuit pack can detect only 12k and 16kHz PPM. If you type 16k in the Frequency field, the circuit pack is set to detect 16kHz.
50	The TN465B, or later, circuit pack and the TN2184 circuit pack can detect only 12k and 16kHz PPM. If you type 50 in the Frequency field, the circuit pack is set to detect 16kHz.
50/12k	The TN465B, or later, circuit pack and the TN2184 circuit pack can detect only 12k and 16kHz PPM. If you type 50/12k in the Frequency field, the circuit pack is set to detect 12kHz.
50/16k	The TN465B, or later, circuit pack and the TN2184 circuit pack can detect only 12k and 16kHz PPM. If you set the Frequency field to 50/16k , the circuit pack is set to detect 16kHz.
2 of 2	

- 7 In the PPM field, type **y**.
- 8 Press **Enter** to save your changes.

Administering PPM for DS1 circuit packs

To administer PPM for DS1 circuit packs:

- 1 Type **change ds1 n**, where **n** is the number of the DS1 circuit pack for which you want to administer PPM. Press **Enter**.

The system displays the **DS1 Circuit Pack** screen ([Figure 74](#)).

Figure 74: DS1 Circuit Pack screen

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: _____

Block Progress Indicator? n

- 2** In the Received Digital Metering Pulse Maximum (ms) field, type the number that your network service provider recommends. Valid entries are **20** to **1000**, in increments of 10. The number that you type in this field must be greater than the number that you type in the Received Digital Metering Pulse Minimum (ms) field.

The system displays this field only when the Signaling Mode field is set to **cas**, the Interconnect field is set to **co** or **pbx**, and the Country Protocol field is set to a protocol that uses PPM as defined in [Table 9, Country protocol codes for incoming digital PPM signaling](#), on page 337.

- 3** In the Received Digital Metering Pulse Minimum (ms) field, type the number the number that your network service provider recommends. Valid entries are **20** to **1000**, in increments of 10. The number that you type in this field must be less than the number that you type in the Received Digital Metering Pulse Maximum (ms) field.

The system displays this field only when the Signaling Mode field is set to **cas**, the Interconnect field is set to **co** or **pbx**, and the Country Protocol field is set to a protocol that uses PPM as defined in [Table 9, Country protocol codes for incoming digital PPM signaling](#), on page 337.

- 4** In the Received Digital Metering Pulse Value field, type the number that your network service provider recommends. Valid entries are **1** and **2**.

The system displays this field when the Signaling Mode field is set to **cas**, the Country Protocol field is set to **21**, and the Interconnect field is set to **co** or **pbx**.

Table 9: Country protocol codes for incoming digital PPM signaling

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

1 of 2

Table 9: Country protocol codes for incoming digital PPM signaling

Code	Country	PPM Min (ms)	PPM Max (ms)	PPM Value
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
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

2 of 2

- 5 Press **Enter** to save your changes.

End-user procedures for Call Charge Information

End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Call Charge Information:

- Displaying call charge information

To display call charge information:

- Press the disp-chrg button before the call drops.
- If you press the:
 - Elapsed-timer button, the elapsed-timer information can overwrite part of the call charge information.
 - Local-directory-number button, the call charge information overwrites the directory number information.
 - Exit or Normal button, the directory number information no longer overwrites the local directory number information.

Reports for Call Charge Information

The following reports provide information about the Call Charge Information feature:

- None

Considerations for Call Charge Information

This section provides information about how the Call Charge Information feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Call Charge Information under all conditions. The following considerations apply to Call Charge Information:

- Performance impact

Call Charge Information can have an impact on system performance in several ways. The information that comes 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 telephones that display charge information and the frequency of updates also affect performance. Usually, the update frequency matches the average rate at which call charge updates are received from the public network.



CAUTION:

When users update displays too frequently, unnecessary system performance degradation can occur. If performance slows to an unacceptable rate, you can lengthen the amount of time between updates.

Interactions for Call Charge Information

This section provides information about how the Call Charge Information feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Call Charge Information in any feature configuration.

- Attendant Features

Attendant consoles cannot have an 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 transfers an outgoing call, the display returns to normal mode. If the transfer is not completed, or the call remains at the attendant console, 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 the disp-chrg again to view call charges. Or if automatic charge display is enabled, the user must wait for the Charge Display Update Frequency interval to expire.

- **Bridged Appearance**

If a user uses a bridged call appearance to place a call, the system displays the call charges on the telephone from which the call is made. If Automatic Charge Display is part of the Class of Restriction (COR), for that telephone, the system displays the charges automatically. The system displays the actual charge for the call on the CDR report as if the call was made from the principal extension, and not from the bridged appearance.

- **Call Coverage or Call Forwarding**

Call charges for a call to an extension that the system redirects over a public-network trunk, are charged to the called extension, not the calling extension. However, if the call is placed from an internal telephone that has charge display capability, the caller sees 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 the disp-chnrg button. The display also shows the charges if the COR of the user allows Automatic Charge Display. If call splitting is enabled, the display shows the charges that accumulated since the user unparked the call.

- **Call Transfer**

Advice of Charge (AOC) administration for the outgoing trunk group controls whether AOC information is requested or recorded for the call, when the system routes a transferred call over a public-network ISDN-PRI trunk group. If two or more outgoing trunks are connected through trunk-to-trunk transfer, the software can receive AOC information from the network for each outgoing trunk that is involved in the call.

- **Call Detail Recording (CDR) Adjuncts**

The software does not tandem AOC information through a private network to other switches. The CDR adjunct that records AOC information must receive its input from software that is directly connected to the public network.

- **CDR Call Splitting**

- The system generates a separate call record, whenever the system transfers a call, if you administered CDR Call Splitting for outgoing trunks.
- Attendant Call Recording, which is 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, request call charge information during the call. However, if you use AOC, a request for call charge information during a call increases message activity on the signaling channel, and reduces Busy Hour Call Capacity.
- In some countries, or with specific protocols, AOC information is unavailable during a call. If AOC information is unavailable during a call, you can use the Elapsed Time in the CDR records to allocate the charges among the call participants.
- You must use CDR Call Splitting if you want the charge display to restart at zero 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 that is in charge-display mode, the display returns to normal. The system does not display call charges when more than two parties are 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 the disp-chrg button again to view call charges. If the automatic charge display is enabled, the user must wait for the system to refresh the display.
- **Last Number Dialed**
When a user is active on a call, a user can view the last number that the user dialed. To view the last number that was dialed, the user presses the stored-numb button, and then presses the last-numb button. To view call charges again, the user must press the disp-chrg button, or the Normal button if Automatic Charge Display is part of the COR of the user.
- **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 stops operating. To resume call charge updates, users must press the Normal button.

Call Coverage

Use the Call Coverage feature to automatically reroute incoming calls to alternate telephone numbers.

Call Coverage supports the following capabilities:

- **Call Coverage**
The system reroutes incoming calls to alternate telephone numbers when the called party is unavailable to answer calls.
- **Call Coverage Off Network**
You can administer and 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.
- **Call Coverage Time of Day**
You can administer the redirection of calls to different lead coverage paths based on the day of the week and the time of day.
- **Call Coverage Changeable Coverage Paths**
Users can use a feature access code (FAC) to modify coverage points.
- **Consult**
Users can answer a coverage call, and then communicate with the called user without the caller hearing the conversation.
- **Extended User Administration of Redirected Calls**
Users can change their lead coverage path from both on-site and off-site locations.

Detailed description of Call Coverage

This section provides a detailed description of the Call Coverage feature.

Use Call Coverage to:

- Reroute incoming calls to alternate telephone numbers when the called party is unavailable to answer calls
- Establish the order in which calls are redirected to alternate destinations
- Establish up to six alternate destinations for an incoming call
- Establish redirection criteria that govern when the system redirects a call
- Establish multiple coverage paths that the system can select from based on redirection criteria
- Redirect calls based on the time of day
- Redirect calls to a local telephone number or to a telephone number in the public network
- Allow users to change their lead coverage path from both on-site and off-site locations

When a call meets the redirection criteria for a called telephone number, the system attempts to route the call sequentially to one of the points in a coverage path. User can have a maximum of six points in the coverage path. If no coverage points are available, the call might revert to the original called number. If any point in the path is available, the call either rings at the individual telephone, an available member of a coverage group, or queues to the coverage group. Once a call is ringing or queued at any point in a coverage path, the call neither reverts to the original number, nor to the previous coverage point.

A call continues to ring at a coverage point for the interval that is administered for Coverage Subsequent Redirection. At the end of this interval, the system attempts to route the call to any points that remain in the coverage path. If no other point is available to accept the call, the call remains queued, or continues ringing at the current coverage point.

What is a Call Coverage Path?

A call coverage path is a list of one to six alternate answering positions. The system sequentially accesses the coverage points when the called party or called group is unavailable to answer the call.

When a call meets the redirection criteria for a called telephone number, the system attempts to route the call sequentially to one of the points in a coverage path. Users can have a maximum of six points in the coverage path. If no coverage points are available, the call might revert to the original called number.

You can assign a coverage path to any of the following entities:

- An automatic call distribution (ACD) split
- An agent login ID
- A PCOL group
- A Terminating Extension Group (TEG)
- A hunt group
- A telephone that can be either on site or off site

You define the coverage paths and establish the redirection criteria. You can include any of the following entities as points in a coverage path:

- A telephone number
- A voice messaging system
- An announcement
- An attendant group
- A uniform call distribution (UCD) hunt group
- A direct department calling (DDC) hunt group
- An ACD hunt group
- A coverage answer group (CAG)
- A vector directory number (VDN)

Multiple coverage paths

The system can select from multiple coverage paths that you define for a single destination. However, the system uses only one coverage path per call. The system first considers *Coverage Path 1*, the lead coverage path, when the system directs a call to coverage.

When the system redirects a call to coverage, the system checks the lead coverage path to determine whether the coverage redirection criteria of the path match the call status. If the criteria match, the system uses the lead coverage path. If the redirection criteria of the lead coverage path does not match, the system moves in sequence from point to point in the coverage path to find a coverage path with redirection criteria that matches the call status. If the system does not find a match, the call remains at the called extension. Once the system selects a coverage path, that path is used throughout the duration of the call.

Time-of-Day Coverage

Use the Time-of-Day Coverage capability to redirect calls to different lead coverage paths at different times of the day, and on different days of the week.

For example, a user might want incoming calls to go to coverage based on the following schedule:

- To a co-worker at the office during normal business hours
- To an off-network destination, such as home, in the early evening
- To a voice messaging, such as AUDIX, at all other times

To provide the user with the requested coverage, you administer the information in [Table 10, Example of a Time-of-Day coverage table](#), on page 345.

Table 10: Example of a Time-of-Day coverage table

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 CovPath3 (Office)	17:30 CovPath3 (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 CovPath3 (Office)	17:30 CovPath3 (Home)	20:00 CovPath3 (AUDIX)
Friday	00:00 CovPath3 (AUDIX)	08:00 CovPath1 (Office)	17:30 CovPath2 (Home)	20:00 CovPath3 (AUDIX)
Saturday	00:00 CovPath3 (AUDIX)			
Sunday	00:00 CovPath3 (AUDIX)			

The Time-of-Day Coverage table represents time in 24-hour format. Activation times are ascending from the earliest to the latest. The activation times cover the entire day. If you do not assign a lead coverage path to a specific time interval, no coverage exists from that time until the next activation time.

When a call arrives at an extension, the system determines the lead coverage path that is in effect at that time. The system uses the information to redirect the call. If you change call coverage for a user while the user has a call in progress, your changes do not affect the call in progress.

Off-Network Call Coverage

You can use standard remote coverage to an external number to send a call to an external telephone. However, the system does not monitor the call once the call leaves your system. Therefore, if the call is busy or not answered at the external number, the call cannot be directed back to the system.

The Coverage of Calls Redirected Off Net capability, however, allows you to use an external number in a coverage path. The system monitors the call to determine whether the external number is busy or unanswered. If necessary, the system can redirect a call to coverage points that follow the external number. Any coverage point can be an off-network destination. This capability allows a call to follow a coverage path that:

- 1** Starts at the called extension
- 2** Redirects to the home telephone, and if not answered at home
- 3** Returns to the voice mail box of the called extension

Note that the call does not return to the system if the external number is the last point in the coverage path, except when no trunks are available to route the call. In that case, the system attempts to again terminate the call at the original called extension.

When the system redirects an incoming trunk call off the network, a timer is set. The timer prevents other incoming trunk calls from redirecting off the network until the timer either expires or is cancelled. The timer prevents calls that were redirected off the network from being routed back to the original telephone number from the off-network destination. Calls that are routed back to the original telephone number in this situation effectively create a loop that seizes trunks until trunks are no longer available.

The system provides the means of performing call classification on an off-network coverage call to determine the disposition of the call. If the off-network call is carried completely over ISDN facilities to the final destination, ISDN trunk signaling is used to monitor the call. If ISDN trunk signaling is not used to monitor the call, a call classifier port is used to cancel the call.

When the system tries to use a call classifier port to classify an off-network coverage call, the system introduces an unavoidable cut-through delay while the call classifier port attempts to identify an answered call. Neither the originating party nor the answering party hears each other during the 1-second or less delay. A call classifier is attached to all off-network coverage calls, that are made over analog facilities or over ISDN facilities, if the call is interworked to non-ISDN facilities on the public network.

When you enable Coverage of Calls Redirected Off-Net:

- The system monitors off-network calls and returns the calls to the system if the calls are not answered within the administered time interval. Calls also return to the system if the system detects a call progress tone, such as busy or reorder.
- A simulated bridge appearance (SBA) is put on the called extension, and the green lamp flashes. A user can answer the call at the called extension at any time.

- When the system uses a call classifier port to classify the call, the system plays local ringback tone to the caller while the system is classifying the off-network call. The system uses the local ringback tone so that the user does not hear what is happening on the public network. As a result, the calling party might not hear the first few syllables that the answering party speaks.
- If any party on the call is on hold when the system routes the call off the network, the call classifier is removed from the call. The call behaves as if Coverage of Calls Redirected Off-Net is not enabled.
- While an off-network call is undergoing call classification, any party that is not already on the call cannot 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 at an off-network destination, or the call is returned to the system for further call processing, these restrictions are removed.
- If the last point in a coverage path is an off-network destination and no trunks are available to route the call, the system attempts to again terminate the call to the called extension.
- The system has no control over any redirection of the call that might take place at an off-network destination. However, further coverage treatment is provided if the off-network redirection interval expires before the call is answered at an off-network destination.

Call Coverage changeable coverage paths

The changeable coverage path capability allows users to modify the coverage paths using a feature access code (FAC).

Extended User Administration of Redirected Calls

The Extended User Administration of Redirected Calls capability allows users to change the lead coverage path or the call forwarding destination from any on-site or off-site location. This capability is also known as remote access. For more information, refer to the Remote Access feature.

Call coverage criteria

Coverage criteria determine the conditions when the system redirects a call from the called extension to the first position in the coverage path. The Call Coverage feature provides the following coverage criteria:

- **Active**
Redirects calls to coverage immediately when the called extension is active on at least one call appearance. For a telephone with only one appearance or a single-line extension, assign the Busy criterion instead of the Active criterion.
- **Busy**
Redirects calls to coverage when all available call appearances at the called extension are in use. The system redirects an incoming call, other than a priority call, to coverage only when all call appearances are in use.
A Terminating Extension Group (TEG) is considered busy if any telephone in the group is active on a call.

Each telephone in a UCD group or a DDC group must be active on at least one call appearance for the system to redirect a call to coverage. If any telephone in the group is idle, the system directs the call to the idle telephone. If no telephone is available, the call can queue if queuing is provided. If queuing is not provided, the system routes the call to coverage. If the queue is full or all agents are in an AuxWork mode, the system routes the call to coverage. Queued calls remain in the queue for the specified interval.

A call does 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 the calls are unanswered during a specified interval. A call rings for the specified number of seconds, and then the system redirects the call to coverage.

- Cover All Calls

Redirects all incoming calls to coverage. This criterion has precedence over any other previously assigned criterion.

- Send All Calls/Go to Cover

Allows users to activate Send All Calls or Go to Cover as overriding coverage criteria. You must assign this redirection criteria before a user can activate Send All Calls or Go to Cover.

- No Coverage

Occurs when no coverage criteria are assigned. The system redirects calls to coverage only when the call extension activates Send All Calls, or the caller activate Go to Cover.

You can combine Active/Don't Answer and Busy/Don't Answer coverage criteria. Other combinations are either not possible or not useful.

You assign redirection criteria separately for internal and external calls. You can link coverage paths so that you can assign Busy Don't Answer for internal calls, and Active for external calls. Similarly, you can assign Busy/Don't Answer for external calls and No Coverage for internal calls. When you assign No Coverage for internal calls, internal calls remain directed to the called telephone or called group.

All calls that are extended by the attendant are treated as external.

VDN in a call coverage path (VICP)

If you assign a vector directory number (VDN) extension as the last point in a call coverage path, you apply call vectoring functionality to the coverage point. The programmable vector that is associated with the VDN provides flexibility in call handling.

You can program a vector that is 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 at the called extension. The same VDN can also be used to retrieve messages. You can also vary the vector program by split status or time-of-day to provide different types of coverage.

When a redirected call covers to a VDN, the simulated bridged appearance of called extension is removed when vector processing starts.

When covered calls or direct calls are connected to AUDIX or to a messaging split through call vectoring, both the original reason for redirection and the called extension 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 communication system message that is sent to the remote switch with AUDIX includes the original reason for redirection and the called extension.

Coverage answer groups

You can create coverage answer groups that allow two to eight telephones to ring simultaneously when calls are redirected to the group. Anyone in the answer group can answer the incoming call.

Announcement in a coverage path

In general, you should not assign an announcement as a point in a coverage path. When the system redirects a call to an announcement, the system plays the announcement and then drops the call. The system drops the call even if there are additional points in the coverage path.

You might want to use an announcement as the last point in a coverage path. The announcement could inform the caller that there is no one to answer the call and advise the caller to call back at another time. Keep in mind that the system drops the call once the system plays the announcement.

Hunt group in a coverage path

You can assign call coverage for a hunt group. If a hunt group queue is full, a call waits for a specified interval. The system then directs the call to the coverage path. The call coverage point can be another hunt group. A call does not cover to a hunt group if no agents are logged in, or if all agents are in AuxWork mode.

Subsequent redirection interval

The subsequent redirection interval controls the number of times that a call rings at a coverage point before the call moves to the next coverage point. The number of rings that the interval control depends on the type of coverage point. For example, the number of rings is different at a local coverage point and a remote coverage point.

Notifying users when the calls are redirected

You can administer a setting that notifies users when the users have a capability active that might redirect the calls. For example, if send all calls or call forwarding is active for a user, you can administer a setting to play a special dial tone when the user goes off hook.

Caller response interval

The system uses a single, short burst of ringing to inform an internal calling party that the system is redirecting a call to coverage. The Call Coverage tone is followed by an optional period of silence, called the Caller Response interval. This interval allows the calling party time to decide whether to:

- Hang up
- Activate Leave Word Calling
- Activate Automatic Callback
- Activate Go to Cover.

When the user cancels the remaining interval, activating Go to Cover is also cancelled.

Consult

When a user answers a coverage call, the user can communicate with the called user without the caller hearing the conversation. This is called private consultation. To consult privately with the called user, the covering user presses the Transfer button, and then the Consult button. When the covering user presses the Transfer button, and then the Consult button, the system places the caller on hold. The system then establishes a connection between the called user and the covering user.

The covering user can create a conference call among the called user, the covering user, and the caller.

The covering user can transfer the call back to the called user.

The system maintains a Consult call at a Temporary Bridged Appearance, if a Temporary Bridged Appearance is available. If a Temporary Bridged Appearance is unavailable, the system uses any idle call appearance for the Consult call. If an idle call appearance is unavailable, the system denies the Consult call.

Features that override Call Coverage

Some features override Call Coverage criteria. The system checks the criteria of the overriding features before the system checks the coverage criteria. The following features override Call Coverage:

- Call Forwarding All Calls
Call Forwarding All Calls temporarily overrides the redirection criteria if Send All Calls is not active. The system attempts to complete the call at the forwarded-to extension before the system redirects the call to coverage. If the redirection criteria of the called extension are met at the forwarded-to extension, the system redirects the call to the coverage path of the called extension.
- Go to Cover
Users can use Go to Cover to send a call directly to coverage, when the users call an internal extension. The internal calling party activates Go to Cover, and can assign Go to Cover to a telephone.

- **Send All Calls**

Users can use Send All Calls to temporarily direct all incoming calls to coverage, regardless of the coverage criteria that are assigned to their extension. The feature also allows users to temporarily remove their telephones from the coverage path of another user.

A user cannot activate Send All Calls, if Send All Calls is not included in the coverage criteria of the extension.

Send All Calls does not affect TEG calls.

- **Send Term**

Send Term is the TEG equivalent of Send All Calls. Since a TEG cannot be in a coverage path, Send Term applies only to a TEG that is called directly.

Conditions that override Call Coverage

Call Coverage redirects calls from the called extension or the called group to alternate answering positions when certain criteria are met. Sometimes calls are sent back to the called extensions or the alternate destination, even though the redirection or overriding criteria are met.

The following list contains the conditions that cause the system to override Call Coverage.

- If no answering positions are available in the coverage path, the call rings at the called telephone, if possible. If the call cannot ring at the called telephone, the calling party receives busy tone. The calling party receives busy tone, even if the Cover All Calls redirection criterion or the Send All Calls overriding criterion is active.
- If the system redirects a call to a coverage point that is unavailable, the call goes to the next coverage point. The call goes to the next coverage point, regardless of the type of coverage that is administered in the coverage point that is unavailable.
- When UCD and DDC group members are unavailable to answer calls to the group, the calls go to a queue, if queuing is available. The call remains in the queue for the call response interval before the system routes the call according to the coverage path. If no points on the path are available, the call remains in the queue. When neither group queuing nor a coverage point is available, the caller receives a us tone or ringback, depending on the type of trunk that carries the call.

If the redirection criterion is Active or Cover All Calls, a called extension can receive a redirection notification signal when the system routes the call to coverage. The redirection notification signal is a short burst of ringing. You can administer the redirection notification signal option for any extensions in your system.

Redirected calls maintain an appearance on the called telephone, if possible. The call appearance status lamp flashes to indicate an incoming call before the system attempts to redirect the call. When the system redirects the call, the status lamp continues to flash. If the system redirects the call to AUDIX, the lamp goes out. If the call appearance is flashing, a user presses the call appearance button to answer the call. If a covering user answers the call, the called user can bridge onto the call. If a covering user answers the call, the status lamp on the telephone of the called user lights steadily.

- Telephone users use Directed Call Pickup to answer calls that ring at another telephone or calls that alert at a coverage point. A call alerts when a call causes a call appearance on the telephone to flash. Directed Call Pickup allows a user to answer an alerting call from any telephone on the system.

- The system routes the following types of calls to the telephone of the called user until the user activates Go to Cover:
 - Priority calls
 - Dial Intercom calls
 - Automatic Intercom calls

The system gives these calls precedence over the redirection criteria, and seizes the call appearance that is usually reserved for outgoing calls, if no other call appearances are available.

Hardware requirements for Call Coverage

ISDN end-to-end signaling uses the ISDN protocol for the call classification. In all other cases, Coverage of Calls Redirected Off-Net (CCRON) requires the following hardware:

- Call Classifier-Detector and Tone Clock/Call Classifier-Detector circuit packs

CCRON usually requires call classification hardware. Both the Call Classifier-Detector and the Tone Clock/Call Classifier-Detector circuit packs provide tone detection ports, including the capability to do call classification. Each circuit pack has eight ports.

For countries that use the USA tone plan, a Call Classifier-Detector or Tone Clock/Call Classifier-Detector circuit pack is sufficient to provide call classification.

For countries that do not use 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
 - Applications that use call classification ports

CCRON competes with the following applications for ports on the Call Classifier-Detector and Tone Clock with Call Classifier-Tone Detector circuit packs:

- Answer Detection
- Call Prompting
- CallVisor ASAI
- Multifrequency Compelled (MFC) signaling

An insufficient number of ports for call classification can seriously degrade performance, and even prevent users from making telephone calls.

Administering Call Coverage

The following steps are part of the administration process for the Call Coverage feature:

- [Creating a coverage path](#)
- [Assigning a coverage path to a user](#)
- [Defining coverage redirected off-network calls](#)
- [Assigning the telephone numbers for the off-network coverage points](#)
- [Assigning time-of-day coverage](#)
- [Creating a coverage answer group](#)
- [Assigning Internal Alerting](#)

This section describes:

- Any prerequisites for administering the Call Coverage feature
- The screens that are required to administer the Call Coverage feature
- Complete administration procedures the Call Coverage feature

Prerequisites for administering Call Coverage

You must complete the following actions before you can administer the Call Coverage feature:

You must complete the following actions before you can administer the Call Coverage feature:

- Ensure that feature access codes (FACs) for activating and deactivating Send All are available on your system, if you want users to use an FAC for these capabilities.

To ensure that FACs for activating and deactivating Send All are available on your system:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the *Feature Access Codes (FAC)* screen ([Figure 75, Feature Access Code \(FAC\) screen](#), on page 354).

Figure 75: Feature Access Code (FAC) screen

change feature-access-codes
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:

Self Station Display Access Code:

Send All Calls Activation:

Station Firmware Download Access Code:

Station Lock Activation:

Station Security Code Change Access Code:

Station User Admin of FBI Assign:

Station User Button Ring Control Access Code:

Telephone Activation: #*

Terminal Dial-up Test Access Code:

Terminal Translation Initialization Merge Code:

Transfer to Voice Mail Access Code:

Trunk Answer Any Station Access Code:

User Control Restrict Activation:

Voice Coverage Message Retrieval Access Code:

Voice Principal Message Retrieval Access Code: *80

Deactivation:

Deactivation:

Deactivation:

Remove:

Separation Code:

Deactivation:

- 2 Page through the screens until you see the Send All Calls Activation field.
- 3 Type an FAC in the Send All Calls Activation field.
- 4 Type an FAC in the Send All Calls Deactivation field.
For more information, see the “Feature Access Code” feature.
- 5 Press **Enter** to save your changes.

Screens for administering Call Coverage

Screen Name	Purpose	Fields
Call Coverage Answer Group	Establish answer groups.	All
Coverage Path	Establish points in the coverage path.	All
Feature Access Code	Assign feature access codes (FACs) to activate or deactivate coverage-related actions.	- Send All Calls Activation - Send All Calls Deactivation
Hunt Groups	Establish groups of users who answer calls for each other.	All
Remote Call Coverage Table	Assign the telephone numbers of remote coverage points.	All

Screen Name	Purpose	Fields
Station	Define coverage information and button assignments for the called user.	• Coverage Path 1 • Coverage Path 2 • Redirect Notification • Button Assignments for goto-cover and send-calls
	Define button assignments for the user to whom the system redirects the call.	Button Assignments for do not disturb (dn-dst), go to cover (goto-cover), and send all calls (send-calls)
	Define the goto-cover coverage point.	Button Assignment for goto-cover
System-Parameters Call Coverage/Call Forwarding	Enable the Call Coverage Off-Net capability.	Coverage of Calls Redirected Off-Net Enabled
Optional Features	Verify that the Call Coverage Off-Net capability is allowed in your system.	Cvg of Calls Redirected Off-Net
Terminating Extension Group	Define the group of users who can answer a call that is simultaneously alerting at the telephones of the group members.	All
Time of Day Coverage Table	Assign coverage throughout the day and week.	All
Trunk Group: • <i>APLT</i> • <i>ISDN-PRI</i> • <i>Tie</i>	Specify the internal ringing and call coverage used for incoming trunk calls.	Internal Alert?

Creating a coverage path

Prerequisites

You must complete the following actions before you can create a coverage path:

- Verify that the settings on the **System-Parameters Call Coverage/Call Forwarding** screen, contain the values that you want for your system. For basic Call Coverage, Avaya recommends that you not change the default settings. However, if you decide to change the default settings, read the field definitions and the field descriptions carefully before you make changes.

To view this screen, type **display system-parameters call coverage/call forwarding**. Press **Enter**.

For a complete description of the **System-Parameters Call Coverage/Call Forwarding** screen, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

To create a coverage path:

- 1 Type **add coverage path next**. Press **Enter**.

The system displays the **Coverage Path** screen ([Figure 76, Coverage path screen](#), on page 356), that shows the next undefined coverage path. The Coverage Path Number field is a display-only field. The total number of coverage paths cannot exceed 600.

Figure 76: Coverage path screen

COVERAGE PATH				
Coverage Path Number: 1		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 Appearances? n				
Point1: 4101	Rng: 3	Point2:	Point3:	
Point4:		Point5:	Point6:	

To see the extensions or groups that use a specific coverage path, type **display coverage sender group n**, where **n** is the coverage path number. For example, you might want to see which extensions use a coverage path before you make changes to the coverage path.

- 2 In the **Hunt After Coverage?** field, type **y** if you want the system to attempt station hunting from the last coverage point, when the coverage point is a busy station. If you do not want the system to attempt station hunting from the last coverage point when the coverage point is a busy station, leave the default set to **n**.
- 3 In the **Next Path Number** field, type a coverage path number in the field if you want the system to redirect if the coverage criteria of the current path does not match the call status. If the coverage criteria of the next path matches the call status, the system uses the coverage criteria to redirect the call, and no other path is searched. If you do not want the system to redirect the call, leave the field blank.

Linkage is a display-only field that shows one or two assigned coverage paths that are linked to the number in the **Next Path Number** field.

- 4** Find the **COVERAGE CRITERIA** area. Note that there is a column for inside calls and a column for outside calls. You can accept the defaults for both columns or only one column. Likewise, you can change the defaults for both columns or only one column. Perform any of the following actions:
- In the **Active?** fields, perform one of the following actions:
 - a** Accept the default **n**, if you *do not* want the call to go to coverage if only one call appearance is busy.
 - b** If you want the call to go to coverage if only one call appearance is busy, type **y**.
 - In the **Busy?** fields, perform one of the following actions:
 - a** Accept the default **y**, if you want the call to go to coverage if the extension is busy.
 - b** If you *do not* want the call to go to coverage if the extension is busy, type **n**.
 - In the **Don't Answer?** fields, perform one of the following actions:
 - a** Accept the default **y**, if you want the call to go to coverage if the number of rings exceeds the number specified in the **Number of Rings** field.
 - b** If you *do not* want the call to go to coverage if the number of rings exceeds the number specified in the **Number of Rings** field, type **n**.
 - In the **Number of Rings** field, type a number from **1** to **99**. This number indicates the number of times a call rings at a telephone before the system redirects the call to the first coverage point. The default is **2**.
 - In the **All?** fields, perform one of the following actions:
 - a** Accept the default **y**, if you want the users with this path to answer their own calls.
 - b** If you *do not* want the users with this path to answer their own calls, type **n**. If you type **n**, the users can never answer their own calls. These user calls always immediately go to coverage.
 - In the **DND/SAC/Goto Cover?** fields, perform one of the following actions:
 - a** Accept the default **y**, if you want users to activate Send All Calls, temporarily direct all incoming calls to coverage (regardless of the assigned Call Coverage redirection criteria), and to temporarily remove their telephone from the coverage path.
 - b** If you *do not* want users to activate Send All Calls, temporarily direct all incoming calls to coverage (regardless of the assigned Call Coverage redirection criteria), and to temporarily remove their telephone from the coverage path, type **n**.
 - In the **Terminate to Coverage Pts. with Bridged Appearance?** field, perform one of the following actions:
 - a** Accept the default **n**, if you want a call to skip the coverage point if the call has already alerted as a bridged call.
 - b** Type **y** to allow a call to alert as a bridged call and a redirected call.

- 5 In the **Point** fields, type the extensions, the hunt group number, or the coverage answer group numbers that you want for coverage points. When you type a number and move to the next **Point** field, the system displays the **Rng** field.
 - Perform one of the following actions:
 - a If you want to use the number of rings entered in the **Number of Rings** field, leave the **Rng** field blank.
 - b If you *do not* want to use the number of rings entered in the **Number of Rings** field, type the number of rings for this coverage point.

NOTE:

To enter an extension that is assigned as a vector directory number (VDN) as the last point in the coverage path, you must make an administration change. For more information, see *Avaya Communication Manager Contact Center Call Vectoring and Expert Agent Selection (EAS) Guide*.

- 6 Press **Enter** to save your changes.

Assigning a coverage path to a user

Prerequisites

You must complete the following actions before you can assign a coverage path:

- Administer the coverage path. For the procedure, see [Creating a coverage path](#) on page 355.

To assign a coverage path to a user:

- 1 Type **change station *n***, where ***n*** is the telephone number of the extension to which you want to assign a coverage path. Press **Enter**.

The system displays the **Station** screen for the extension that you requested ([Figure 77](#)).

Figure 77: Station screen

STATION		
Extension: 30019	Lock Messages? n	BCC: 0
Type: 4612	Security Code: *	TN: 1
Port: S04007	Coverage Path 1:	COR: 1
Name: station 30019	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 30019	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english		
	Media Complex Ext:	
	IP SoftPhone? n	

- 2 Page through the screens until you see the **Coverage Path 1** field. Type a coverage path number of a previously administered Call Coverage path screen.
 - Perform one of the following actions:
 - a In the **Coverage Path 2:** field, type a coverage path number of a previously administered Call Coverage path screen, if you want the extension to have an alternative coverage path.
 - b If you *do not* want a second coverage path, leave the field blank.
- 3 Press **Enter** to save your changes.
- 4 Page through the screens until you find the **Redirect Notification?** field ([Figure 78, Station screen](#), on page 359).
 - Perform one of the following actions:
 - a Accept the default **y** if you want a half ring at the telephone when the system redirects a call to coverage.
 - b If you do not want the half ring, type **n**.

Figure 78: Station screen

FEATURE OPTIONS	STATION
LWC Reception: 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	Restrict Last Appearance? y
Bridged Call Alerting? n	
Active Station Ringing: single	
H.320 Conversion? n	Per Station CPN - Send Calling Number?
Service Link Mode: as-needed	Audible Message Waiting? n
Multimedia Mode: enhanced	Display Client Redirection? n
MWI Served User Type:	Select Last Used Appearance? n
AUDIX Name:	Coverage After Forwarding? s
	Multimedia Early Answer? n
	Direct IP-IP Audio Connections? y
	IP Audio Hairpinning? y
Emergency Location Ext: 30019	

- 5 Press **Enter** to save your changes.

- 6 Page through the screens until you find the **BUTTON ASSIGNMENTS** area ([Figure 79, Station screen](#), on page 360). If you want the user to have buttons on the telephone for do not disturb, go to cover, or send all calls, use this area to assign the buttons. Note that you can use assign any, all, or none of the buttons, but you can make only one assignment per button. Move to the button that you want to use, and perform any of the following actions:
 - a If you want to assign a do-not-disturb button, type **dn-dst** after a button number.
 - b If you want to assign a go-to-over button, type **goto-cover** after a button number.
 - c If you want to assign a send-all-calls button, type **send-calls** after a button number. When you click **next page** or press **Tab** or **Enter**, the system displays the **Ext** field. If you want to send calls to the extension you specified when you typed the **change station** command, leave the **Ext** field blank. If you want to send calls to another extension, type the extension number to which the system redirects calls when the user presses the send-calls button.

Figure 79: Station screen

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	7:
2: call-appr	8:
3: call-appr	9:
4:	10:dn-dst
5:	11:goto-cover
6:	12:send-calls Ext:

- 7 Press **Enter** to save your changes.

Assigning a Consult button for a user

To assign a Consult button for a user:

- 1 Type **change station n**, where **n** is the extension of the user to whom you want to assign the Consult capability. Press **Enter**.
The system displays the **Station** screen for the extension that you requested ([Figure 79, Station screen](#), on page 360).
- 2 Page through the screens until you see the **BUTTON ASSIGNMENTS** area.
- 3 In the **BUTTON ASSIGNMENTS** area, type **consult** next to the button that you want the user to use for Consult.
- 4 Press **Enter** to save your changes.

Defining coverage redirected off-network calls

To define coverage for calls that the system redirects to external, off-network calls, you must complete the following procedures.

- Assigning the telephone number for the external coverage point
- Administering the coverage path for calls redirected to external numbers

Prerequisites

You must complete the following actions before you define coverage for calls that the system redirect off the network:

- On the *Optional Features* screen, verify that the **Cvg of Calls Redirected Off-Net** field is set to **y**. To view this screen, type **display system-parameters customer-options**. Press **Enter**. If the **Cvg of Calls Redirected Off-Net** field is set to **n**, your system is not enabled for the Call Coverage Off Network capability. Contact your Avaya representative for assistance before continuing with this procedure.

For a complete description of the *Optional Features* screen, [click here](#) or see the *Administrator's Guide for Avaya Communication Manager*.

Assigning the telephone numbers for the off-network coverage points

To assign the telephone numbers for the off-network coverage points:

- 1 Type **change coverage remote n**, where **n** is the number of the Remote Call Coverage table that you want to change. Valid numbers are between 1 and 10. Press **Enter**.

The system displays the *Remote Call Coverage Table* ([Figure 80](#)).

Figure 80: Remote Call Coverage Table screen

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 the telephone number (maximum of 16 digits) of the remote coverage point in one of the remote call coverage table fields. If you need a digit to get outside your network, add the digit before the external number. In this example, the system requires a **9** to place outside calls. The sequentially numbered fields in which you assign telephone numbers are called *remote code numbers*. You need this number to complete the procedure for defining coverage for calls redirected to external numbers. In this example the remote code number is **r1**. The letter **r** indicates remote, and **1** refers to the number 01 where you typed the telephone number on the screen. If you type a telephone number at number 15 on the screen, the remote code number for that entry is **r15**.
- 3 Press **Enter** to save your changes.

Administering the coverage path for redirected off-network calls

- 1 Type **change coverage path coverage path number**, where *coverage path number* is the number that is assigned to the coverage path that you want to administer for off-network coverage. Press **Enter**.

The system displays the **Coverage Path** screen ([Figure 81, Coverage Path screen](#), on page 362).

Figure 81: Coverage Path screen

COVERAGE PATH					
Coverage Path Number: 1			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 Appearances? n					
Point1: 4101	Rng: 3	Point2: r1	Rng:	Point3: h77	Rng:
Point4:		Point5:		Point6:	

To enter an extension that is assigned as a vector directory number (VDN) as the last point in the coverage path, you must make an administration change. For more information, see *Avaya Communication Manager Contact Center Call Vectoring and Expert Agent Selection (EAS) Guide* for more information.

- 2 In the **Coverage Points** area, type the remote code number. The **Point** field that you select, determines where the number is used in the coverage path. In this example:
 - **r1** is coverage point 2.
 - A call first covers to extension 4101.
 - If extension 4101 is unavailable or busy, the system redirects the call to the off-network number that is assigned to remote code number **r1** in the Remote Call Coverage table.

When you move to the next **POINT** field, the system displays the **Rng** field. If you want to use the number of rings displayed in the **Number of Rings** field on this screen, leave the **Rng:** field blank.

If you *do not* want to use the number of rings displayed in the **Number of Rings** field on this screen, type the number of rings for this coverage point.

- 3 Press **Enter** to save your changes.

Assigning time-of-day coverage

To assign time-of-day coverage for a user, you must complete the following procedures:

- Setting up a time-of-day coverage plan
- Assigning a time-of-day coverage plan to a the extension of the user

Prerequisites

You must complete the following actions before you can define a time-of-day coverage path:

- Administering the coverage path
- Setting up a time-of-day coverage plan

To administer the coverage path, see [Creating a coverage path](#) on page 355.

To set up a time-of-day coverage plan:

- 1 Type **add coverage time-of-day next**. Press **Enter**.

The system displays the next **Time of Day Coverage Table** screen ([Figure 82, Time of Day Coverage Table](#), on page 363). If this is the first Time of Day Coverage plan in your system, the table number is 1. This is the table number that you assign to a user extension.

Figure 82: Time of Day Coverage Table

TIME OF DAY COVERAGE TABLE: 1										
	Act Time	Cvg Path	Act Time	Cvg Path	Act Time	Cvg Path	Act Time	Cvg Path	Act Time	Cvg Path
Sun	0:00	3	:	:	:	:	:	:	:	:
Mon	0:00	3	08:00	1	17:30	2	20:00	3	:	:
Tue	0:00	3	08:00	1	17:30	2	20:00	3	:	:
Wed	0:00	3	08:00	1	17:30	2	20:00	3	:	:
Thu	0:00	3	08:00	1	17:30	2	20:00	3	:	:
Fri	0:00	3	08:00	1	17:30	2	20:00	3	:	:
Sat	0:00	3	08:00	1	17:30	2	20:00	3	:	:

- 2 To define your coverage plan, type the time period and the path number for each day of the week that you want to cover.

Enter the time in a 24-hour format, from the earliest to the latest. For this example, assume that coverage path 1 goes to the co-worker, coverage path 2 goes to the home, and coverage path 3 goes to voice mail. In this example, the user has the following coverage:

- During the work day from 08:00 to 05:29, the system uses coverage path 1 to route calls to a co-worker.
- In the evening from 05:30 to 19:59, the system uses coverage path 2 to route calls to home.
- At night from 20:00 to 24:00, the system uses coverage path 3 to route calls to voice mail.

Define the path for the time period from 00:01 to 23:59 that you want coverage to operate. If you do not assign a coverage path to a specific time interval, no coverage exists from that time until the next coverage path activation time.

- 3 Press **Enter** to save your changes.

To assign the time-of-day coverage to a user.

- 1 Type **change station *n***, where *n* is the user telephone extension number. Press **Enter**.

The system displays the **Station** screen ([Figure 83, Station screen](#), on page 364).

Figure 83: Station screen

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 In the Coverage Path 1 field, type **t** plus the number of the Time of Day Coverage Table. In this example, the coverage path is **t1**. The system redirects calls to this extension as defined in the Time of Day Coverage Table number 1.
- 3 Press **Enter** to save your changes.

Creating a coverage answer group

To add a coverage answer group:

- 1 Type **add coverage answer-group** next. Press **Enter**.

The system displays the next *Coverage Answer Group* screen ([Figure 84](#)).

Figure 84: Coverage Answer Group

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, type a name to identify the coverage group. The default name is **COVERAGE GROUP**.
- 3 In the **Ext** column, type the extension of each group member that you want to include in this group. You can type only one extension in each **Ext** field.
- 4 Press **Enter** to save your new group list. The system automatically completes the **Name** field when you press **Enter**.

Assigning Internal Alerting

To assign internal alerting for APLT, ISDN-PRI, or Tie trunks:

- 1 Type **change trunk-group *n***, where *n* is the number of the trunk-group for which you want to administer Remote Access.

The system displays the **Trunk Group** screen ([Figure 85](#)).

Figure 85: Trunk Group screen

change trunk-group 1		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	
	Modify Tandem Calling Number?: n	
Send UCID? n	BSR Reply-best DISC Cause Value: 31	
Send Codeset 6/7 LAI IE? y		
Path Replacement with Retention? n		
Path Replacement Method: better-route		
SBS? y		

- 2 Page through the screens until you see the **Internal Alert?** field.
- 3 In the **Internal Alert?** field, perform one of the following actions:
 - If you want internal ringing and coverage for your system, type **y**.
 - If you do not want internal ringing and coverage for your system, type **n**.
- 4 Type **Enter** to save your changes.

Reports for Call Coverage

The following reports provide information about the Call Coverage feature:

- The Coverage Path Measurement report shows coverage activity about the coverage paths.
- The Principal Coverage Measurement report shows coverage activity about the called extensions.
- The Call Detail Recording (CDR) report shows the outgoing trunk calls.

For more information on these reports and the associated commands, see *Reports for Avaya Communication Manager*.

Considerations for Call Coverage

This section provides information about how the Call Coverage feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Call Coverage under all circumstances.

- Tie-trunk calls

Incoming tie-trunk calls can be administered as either internal calls or external calls, and are redirected to Call Coverage accordingly.

Interactions for Call Coverage

This section provides information about how the Call Coverage feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Call Coverage in any feature configuration.

- Agent Call Handling

Do not assign Cover All Calls to agents who have the Automatic Answer option enabled. Any Automatic Call Distribution (ACD) or any non-ACD call to an extension on which Automatic Answer is enabled, and has its coverage redirection criteria administered as Cover All Calls, does not go to coverage. Instead, the call goes to the called extension. Cover All Calls redirection criteria do not affect incoming calls when a user is in the Auto-Answer mode.

- Answer Detection

Coverage of Calls Redirected Off-Net (CCRON) competes with Answer Detection for call classifier ports.

- Automatic Callback and Ringback Queuing

The system does not redirect callback calls to coverage. The caller can activate Automatic Callback when the user hears a ringing, redirection notification signal or a busy signal.

- Automatic Intercom, Dial Intercom, and Priority Calling

The system does not redirect calls that use these features 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 that is associated with the bridged call appearance.

If a telephone user activates 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.

While an off-network call is undergoing call classification, the system blocks a user from bridging onto the call.

- Call Detail Recording (CDR)

When the Coverage of Calls Redirected Off-Net field is enabled, the system generates a CDR record only after the call is answered off the network. The dialed number in the record is the off-network number to which the call covers. The calling number is the station that is covered to the off-network destination.

- **Call Forwarding**

Call Forwarding temporarily overrides the redirection criteria. When the redirection criteria are met at the forwarded-to extension, the system redirects the call to the coverage path of the forwarding extension.

The system allows calls that are forwarded off the network to be tracked for busy or no-answer conditions, and to return for further call-coverage processing under those conditions. However, if the called extension does not have a coverage path, the system does not track the call and the call is left at the off-network destination, regardless of whether the call is answered or busy.

If both Send All Calls and Call Forwarding are active, the system immediately redirects most calls to that extension to coverage. However, the system forwards priority calls 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, the system immediately directs most calls to that extension to coverage. However, the system forwards priority calls to the designated forwarding destination.

Activation of Send All Calls at the forwarded-to extension does not affect calls that are forwarded to that extension.

- **Call Pickup**

Any call that the system directs 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 (CCRON) competes with the Call Prompting feature for call classifier ports.

- **CallVisor ASAI**

Coverage of Calls Redirected Off-Net (CCRON) competes with CallVisor for call classifier ports.

- **Centralized Attendant Service (CAS)**

If an incoming CAS call is directed to a hunt group, the call is not redirected to the coverage path of the hunt group.

- **Class of Restriction (COR) and Controlled Restrictions**

Users who might usually be restricted from receiving calls can receive calls that the system directs to them from the Call Coverage feature.

- **Conference**

The system blocks users from conferencing another party onto a call that was routed off the network while the call is undergoing call classification. If any party on the call is on hold, the system routes the call off the network, but the system does not attempt to classify the call. The system routes the call off the network, even when the Coverage of Calls Redirected Off-Net field is enabled.

A call that covers to a vector directory number (VDN) cannot be added to a conference while the call is in vector processing.

- **Direct Department Calling (DDC), Uniform Call Distribution (UCD), and Automatic Call Distribution (ACD)**

If a user with an Auxiliary Work button activates or deactivates Send All Calls, the Auxiliary Work function that is associated with the DDC feature or the UCD feature is activated or deactivated simultaneously.

If a user has no Auxiliary Work button, activating or deactivating Send All Calls makes the user unavailable or available, respectively, for DDC and UCD calls, but Auxiliary Work is not activated or deactivated. The user can use a feature access code (FAC) to activate or deactivate Auxiliary Work mode.

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 (CCRON) competes with DOD for call classifier ports when DOD uses 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, because Non-MFC DOD calls do not need MFC tones.

- **Global Call Classification**

To classify tones in countries that do not use the USA tone plan, time cadences and frequencies must be administered so that time cadences and frequencies 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 called user picks up on the call, the coverage appearance might be dropped, depending on administration.

If any party is on hold when the system routes a coverage call off the network, that call does not undergo call classification. In this case, the call does not undergo call classification, even when the Coverage of Calls Redirected Off-Net field is enabled on your system.

- **Internal Automatic Answer (IAA)**

If call coverage redirection criteria redirects an internal call to another telephone, that call is eligible for IAA at that telephone.

IAA does not apply to calls to the original called extension when:

- The user at the called extension has Do Not Disturb, Send All Calls, or Cover All Calls active
- The calling user selects Go To Cover before the user places the call

Calls that are directed to a Coverage Answering Group cannot use IAA.

- **ISDN End-to-End Calls**

When ISDN facilities carry an off-network coverage call end-to-end, call classification is accomplished through the ISDN protocol, rather than by a call classifier port.

- **Leave Word Calling (LWC)**

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 called user to call the covering user. When a covering user activates Coverage Callback during a coverage call, a message is left for the called user to call the internal caller.

- **Tenant Partitioning**

The Tenant Partitioning feature might not block coverage calls across tenant partitions.

Troubleshooting Call Coverage

This section lists the known or common problems that users might experience with the Call Forwarding feature:

Problem	Possible cause	Action
The system redirects some unanswered calls to the Attendant console, rather than to the coverage path.	A call that is transferred internally, and not answered within the Return Call Timeout interval on the Console Parameters screen, redirects to the attendant console.	Verify that the called extension has a coverage path by typing the status station command. If the active coverage option field on the Station screen has a coverage path, increase the number of seconds in the Return Call Timeout interval on the Console Parameters screen. Increase the interval so that it exceeds the total time that a call rings on the called extension and all points in the coverage path of the called extension.
Calls are not redirecting to the correct destination in the coverage path of the called extension.	The called extension might be forwarded to another destination.	Type the status station command to determine if either Call Forwarding or Send All Calls is active at the called extension. If the CF Destination Ext field contains a forwarded-to extension, cancel Call Forwarding for the called extension. If the SAC Activated? field is set to yes , cancel Send All Calls for the called extension.
	The coverage path that you assigned to the called extension might be incorrect.	Type the display station command to verify that you assigned the correct coverage path to the called extension. Change the coverage path if it is not the coverage path that you want for the called extension.

Problem	Possible cause	Action
Calls are not redirecting to the coverage path of the called extension.	The points in the coverage path might not be available.	Type the status station command to determine if either Call Forwarding or Send All Calls is active at the called extension. If the CF Destination Ext field contains a forwarded-to extension, cancel Call Forwarding for the called extension. If the SAC Activated? field is set to yes, cancel Send All Calls for the called extension. If the coverage path of the called extension contains a hunt group, ensure that the length of the queue is sufficient to contain all the calls that redirect to the hunt group.

Call Detail Recording

Use the Call Detail Recording (CDR) feature to record information on incoming, outgoing, and tandem calls for each trunk group, that you administer for CDR, including auxiliary trunks. The system records information on each trunk-group call and each station-to-station call.

Call Detail Recording supports the following capabilities:

- **Account Code Dialing**
You use the Account Code Dialing capability to associate a call with an account number.
- **Forced Entry of Account Codes (FEAC)**
Use the FEAC capability to require that users enter an account code when the users call a telephone number. In this case, the system rejects any calls that do not have an accompanying account number.
- **Call Splitting**
Use the Call Splitting capability to record information about calls that:
 - Are part of a conference
 - Are transferred
 - Involve an attendant
- **Intraswitch CDR**
Use Intraswitch CDR to record information about calls that are to and from users on your system.
- **CDR Privacy**
Use the CDR Privacy capability to maintain the privacy of the caller. The system replaces some of the digits that the user dials with blanks. Thus, the system records the call information, including the account number that the user enters, but the CDR information does not show the telephone number that the user dials.

Detailed description of Call Detail Recording

Use the Call Detail Recording (CDR) feature to record information on incoming, outgoing, and tandem calls for each trunk group, that you administer for CDR, including auxiliary trunks. The system records information on each trunk-group call and each station-to-station call.

You can also request that CDR record information on:

- Temporary signaling connections (TSCs) that involve trunks
- Calls that use loudspeaker paging
- Calls to which account code dialing or a Feature Access Code (FAC) apply

- Ineffective call attempts

If you request that the system record information in ineffective call attempts, you greatly increase the number of calls that the system records. However, the request to record ineffective call attempts can also help you to increase security, because the system records call attempts that are blocked because of insufficient calling privileges.

Information on ineffective call attempts can also show you that your users cannot make calls because all the trunks on your system are busy.

- The audio service link calls that the switch uses for IP softphones that are set up as telecommuter IP softphones

An IP softphone can use one audio service link to make many short calls. The system shows these many short calls as one long call on the CDR reports.

Some call accounting systems do not support all the information that CDR offers. See your Avaya representative for information on how CDR operates on your system.

Monitor call detail records

Monitor call detail records daily for unusual calling patterns, long calls, international calls, calls that are outside the normal business hours, and other indications of toll fraud. Call accounting systems are available that automatically monitor CDR output for fraudulent calling patterns.

Answer detection

Avaya Communication Manager provides three methods to determine whether the called party answers a call:

- Call classification
- Network answer supervision
- Answer supervision by timeout

Call classification

A call-classifier circuit pack detects tones and voice-frequency signals on the line to determine whether a call is answered. This method is fairly accurate. The calls that are answered are usually classified correctly. But the following exceptions exist:

- Miscellaneous tones, such as confirmation tones, might be classified as answers.
- Loud background noise might activate answer detection, and cause the a call to be classified as answered, even if the call is not connected.
- Some calls that are answered might be incorrectly classified as fast busy signals.
- Call classifier circuit packs do not recognize Special Information Tones (SIT) as answers.

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 the call is subsequently answered, the call duration that CDR reports includes both the time between the incorrect classification and the actual answer, and the remaining duration of the call.

Network Answer Supervision

The central office (CO) sends a signal to the originating switch when the far end answers a call. If a call travels over a private network before the call reaches the CO, the signal is transmitted back over the private network to the originating switch. This method is extremely accurate, but it is unavailable over most loop-start trunks. For example, network answer supervision is not available over central office (CO), foreign exchange (FX), and Wide Area Telecommunications Service (WATS) trunks in the United States.

Answer Supervision by Timeout

If the caller is off-hook when the answer timer expires, the system assumes that the outgoing call is answered. Answer supervision by timeout is the least accurate method to detect that a call is answered. Calls that are shorter than the timer duration do not generate call records. Calls that ring for a long time produce call records, even if the calls are unanswered.

Network Answer Supervision overrides Answer Supervision by Timeout.

Account Code Dialing

Use the Account Code Dialing capability to associate a call with an account number. A user enters a feature access code (FAC) for Account Code Dialing before a user dials a telephone number. You can specify that the use of the FAC is mandatory or optional for the user. When a user dials a telephone number and the FAC, the system records the:

- Telephone number
- Account code
- Trunk Access Code (TAC), or the Automatic Route Selection (ARS) access code

The system does not record the FAC for Account Code Dialing.

Forced Entry of Account Codes

If you require that users enter an account code FAC, you have several options to consider. You can require that:

- All users enter an account code for all calls
- All users enter an account code for calls made on a specific trunk
- All users enter an account code for calls to a specific telephone number
- A specific user enters an account for all calls made by that user

If you use the FEAC capability, the system rejects any call that a user makes without an account code FAC, if the call requires an account code. When the system rejects the call, the user hears intercept tone.

Avaya recommends that you use the FEAC capability to make your system more secure.

SECURITY ALERT:

The system does not verify account codes. The system only verifies that the user enters the number of digits that you specify. If you want the system to verify account codes, you need to use the Authorization Codes feature. For more information on the Authorization Codes feature, see the *Administrator's Guide for Avaya Communication Manager*.

The following types of calls never require an account code:

- Calls made by an attendant
- Calls that an attendant makes to determine if a trunk is busy
- Calls that a user makes to determine if a trunk is busy
- Distributed Communications System (DCS) calls, unless the Class of Restriction (COR) of the trunk requires an account code
- Personal Central Office Line (PCOL) calls
- Remote access calls that do not have barrier codes
- Trunk-to-trunk calls

Call Splitting

You use the Call Splitting capability to record information about the following calls:

- Are part of a conference
- That transferred
- That involve an attendant

The system records a separate CDR record for each participant on any of these types of calls.

You can request call splitting information for both incoming and outgoing trunks. You can also request call splitting information for an attendant call on an incoming trunk, and for an attendant call on an outgoing trunk

Incoming trunk call splitting

If you request incoming trunk call splitting (ITCS) information for a call, the system creates a CDR record when a user uses the Conference feature or the Transfer feature. The CDR record includes the:

- Duration of the user participation
- Incoming trunk access code
- Number that the caller dialed
- Condition code

When a user drops a call, or successfully transfers a call, the system records the action of the user. The duration of a transferred call starts when the transferring party presses the transfer button for the second time.

When a user uses the Conference feature for an incoming trunk call, the system creates a CDR record when the user adds a participant to the conference call. The CDR record of these calls shows the duration of the call for each user who participated. The CDR records of a conference call contain duration information that overlaps.

The system creates an incoming trunk call record when:

- You requested ITCS for your system.
- A user adds another user to a conference call. or a user transfers another user
- The user who is added to the conference call, or who is transferred, is on a local extension that has the Intraswitch CDR option activated.

The system does not create an Intraswitch CDR record.

An example of ITCS and a conference call

The following example shows the interaction between the participants of a conference call and the system, when ITCS is active, and all participants are on the same server:

- Caller A, at extension 123, makes an incoming trunk call to participant B, at extension 565-7890.
- Caller A and participant B talk for 2 minutes.
- Participant B adds participant C, at extension 5-4321, to the conference call.
- Participant B adds participant D, at extension 5-9876, to the conference call.
- Caller A, participant B, participant C, and participant D talk for an additional 8 minutes.
- Participant B drops the call.
- The system creates a CDR record for call segment A–B.
- Caller A, participant C, and participant D talk for an additional 5 minutes.
- Caller A, participant C, and participant D drop the call.
- The system creates two additional CDR records, one for call segment A–C and one for call segment A to D. Note that each CDR record shows the incoming trunk ID as the calling number, 123.

[Table 11, ITCS conference call on the same server](#), on page 377 shows the CDR information that changes when ITCS is active during a conference call. The call durations are approximate.

Table 11: ITCS conference call on the same server

Call segment	Call 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

An example of ITCS and a call transfer on the same server

The following example shows the interaction between the participants of a transfer call and the system when ITCS is active, and all participants are on the same server:

- Caller A, at extension 123, calls participant B, at extension 565-7890.
- Caller A and participant B talk for 1 minute.
- Participant B transfers the call to participant C, at extension 5-4321.

- The system creates a CDR record for call segment A–B.
- Caller A and participant B talk for an additional 5 minutes.
- Caller A and participant B drop the call.
- The system creates a CDR record for call segment A–C.

[Table 12, ITCS transfer on the same server](#), on page 378 shows the CDR information this scenario. The call durations are approximate

Table 12: ITCS transfer on the same server

Call segment	Call 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

An example of ITCS and a call transfer to the public network

The following example shows the interaction between the participants of a transfer call and the system when ITCS is active, and all the participants are *not* on the same server:

- Caller A, at extension 123 on server, calls participant B, at extension 565-7890.
Both caller A and participant B are on the same server.
- Caller A and participant B talk for 1 minute.
- Participant B transfers the call to participant C, at telephone number 566-5555.
Participant C is on the public network.
- Participant B and participant C talk for an additional 4 minutes.
- Participant B and participant C drop the call.
- The system creates two CDR records, one for call segment A–B, and one for call segment A–C.

[Table 13, ITCS transfer to an outgoing trunk](#), on page 378 shows the CDR information that changes when ITCS is active during a call transfer. The call durations are approximate. Note that the duration of the original incoming trunk call, call segment A to B, includes the duration of the conversation between caller A and participant B, and the duration of the conversation between participant B and participant C.

Table 13: ITCS transfer to an outgoing trunk

Call segment	Call 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 you request outgoing trunk call splitting (OTCS), the system creates CDR records of transferred outgoing calls in the same manner as for ITCS. See [Table 12, ITCS transfer on the same server](#), on page 378 and [Table 13, ITCS transfer to an outgoing trunk](#), on page 378 for a description of the CDR information on transfer calls when ITCS is active.

If you request OTCS, and a user originates a conference call, the call duration for that user starts when the user originates the call. The call duration for that user ends when the user drops the call. When the user drops the call, the system creates a second CDR record for the users who remain on the conference call.

An example of OTCS and a conference call on the public network

The following example shows the interaction between the participants of a transfer call and the system, when OTCS is active, and all the participants are *not* on the same server:

- Caller A, at extension 123 on server, calls participant B, at telephone number 777-7890.
Participant B is on the public network.
- Caller A and participant B talk for 5 minutes
- Caller A adds participant C to the conference call.
- Participant B transfers the call to participant C, at telephone number 777-5678.
Participant C is on the public network.
- Caller A, participant B, and participant C talk for an additional 5 minutes.
- Caller A, participant B, and participant C drop the call.
- The system creates two CDR records, one for call segment A–B, and one for call segment A–C.

[Table 14, OTCS conference call](#), on page 379 shows the CDR information that changes when OTCS is active during a conference call. The call durations are approximate.

Table 14: OTCS conference call

Call segment	Call 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

An example of OTCS and a call transfer to a public network

The following example shows the interaction between the participants of a transfer call and the system, when OTCS is active, and all the participants are *not* on the same server:

- Caller A, at extension 51234 on server, calls participant B, at telephone number 777-7890.
Participant B is on the public network.
- Caller A and participant B talk for 5 minutes.

- Caller A transfers the call participant C, at extension 54444.
Caller A and participant C are on the same server.
- The system creates two CDR records, one for call segment A–B, and one for call segment C–B.

[Table 15, OTCS call transfer](#), on page 380 shows the CDR information that changes when OTCS is active during a call transfer. The call durations are approximate.

Table 15: OTCS call transfer

Call segment	Call 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

ITCS and OTCS and attendant call recording

If you request either ITCS or OTCS, you have the option for the system to generate a CDR record of the attendant portion for calls that are transferred.

If you request either ITCS or OTCS, the system always creates a separate CDR record of the attendant portion of a conference call.

An example of an attendant incoming trunk call transfer

The following example shows the interaction between the participants of a transfer call and the system, when either ITCS or OTCS is active:

- Caller A, at TAC 123, calls the attendant, and asks the attendant to transfer the call to participant B, at extension 5-888
Caller A is on the public network.
The attendant and participant B are on the same server.
- Caller A and the attendant talk for 1 minute.
- Caller A and participant B, talk for 5 minutes.
- The system creates two CDR records, one for call segment A–Attd, and one for call segment A–B.

[Table 16, Attendant transfer of an incoming trunk call](#), on page 381 shows the CDR information that changes when ITCS or OTCS is active when an attendant transfers an incoming public-network call.

Table 16: Attendant transfer of an incoming trunk call

Call Segment	Call 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

An example of an attendant call transfer on a public-network trunk

The following example shows the interaction between the participants of a transfer call and the system, when either ITCS or OTCS is active, and an attendant transfers a call to the public network:

- The attendant dials participant A at extension 5-9999.
- The attendant and participant A talk for 1 minute.
- The attendant transfers the call to participant B at telephone number 444-5678.
- The participant B and participant B, talk for 5 minutes.
- The system creates two CDR records, one for call segment A–Attd, and one for call segment A–B.

[Table 17, Attendant call transfer on a public-network trunk](#), on page 381 shows the CDR information that changes when ITCS or OTCS is active when an attendant transfers a call to an outgoing public-network trunk.

Table 17: Attendant call transfer on a public-network trunk

Call segment	Call 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

Intrswitch CDR

The system uses the Intrswitch CDR capability to create CDR records for calls to and from users on the same local server. Before the system can create an intrswitch CDR record, you must assign the Intrswitch CDR capability for one of the extensions.

If you enable ITCS for your system, and you assign the Intrswitch CDR capability for an extension, the system-wide ITCS overrides the Intrswitch CDR for the extension. When the system-wide ITCS overrides the Intrswitch CDR for the extension, the system generates trunk call records for an incoming trunk call to the extension. The system does not generate Intrswitch CDR records for an incoming trunk call to the extension.

The records that the system creates for the Intrswitch CDR capability are similar to the records that the system creates for other CDR records. However, some of the information differs. For example, the system does not provide trunk access codes (TACs) or circuit IDs for intraswitch calls, because that information is unnecessary.

Some calls might appear to be intraswitch CDR calls, but are actually trunk calls. For example, the system creates a trunk CDR record for an internal call to an extension that is forwarded to an outgoing trunk, even if you assigned the Intraswitch CDR capability for either station.

You can assign the Intraswitch CDR capability to:

- A Terminating Extension Group (TEG)
- A station
- A data module
- A Vector Directory Number (VDN)
- A Primary Rate Interface (PRI)
- An endpoint
- An access endpoint
- A hunt group

The number that appears in the **Dialed Number** field depends on whether you administered the CDR System Parameters to record hunt group/member or VDN information. You *cannot* assign the Intraswitch CDR capability to an attendant console or a CallVisor Adjunct-Switch Application Interface (ASAI) station.

Note that the system generates a CDR record for a call only if the user is the caller or the recipient of the call, and has the Intraswitch CDR capability active for the extension of the user. For example, if the user participates in the call as the result of Call Pickup or Call Forwarding, the system does not create a CDR record.

CDR Privacy

Use the CDR Privacy capability to maintain the privacy of the caller. The system replaces some of the digits that the user dials with blanks. The system records the call information, including the account number that the user enters. But the CDR information does not show the telephone number that the user dials.

You can assign the CDR Privacy capability individually to each of your users. You decide the number of digits that the system replaces with blanks. The system then uses this information for all calls of the users to whom you assign the CDR Privacy capability.

There are some conditions in which the CDR Privacy capability does not apply.

- Some countries require that the system replace a specific number of the digits that the user dials with blanks. If a country requires that the system replace a specific number of the digits that the user dials with blanks, the requirement applies to all calls.
- When an adjunct-originated call is made on behalf of a hunt group and the **Calls to Hunt Group - Record** field on the **CDR System Parameter** screen is set to **group-ext**, then CDR privacy does not apply.
- When an adjunct-originated call is made on behalf of a hunt group and the **Calls to Hunt Group - Record** field on the **CDR System Parameter** screen is set to **member-ext**, then CDR privacy applies.
- Some report processors do not support the CDR Privacy capability.

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 usually used for a local storage format call detail recording utility (CDRU) to provide CDR data to the Network Control Operations Support Center (NCOSS) to assess network performance or helping to find network problems.

The primary and secondary CDR output device ports are independent of each other. Each port works even if the link to the other port is not operating. If a link to a CDR output device port is not operating for more than one minute, the system might lose some data. However, the system stores the most recent CDR records for the primary CDR output device, even when some records are lost. The system stores the most recent 500 CDR records for Release 5vs/si/csi and later, and the most recent 1,900 CDR records for Release 5r and later. When the system restores the link, the system sends the CDR records to the CDR output device on a first-in, first-out basis.

You can decide how the system responds when the CDR buffer is full. You can administer the system to do one of the following:

- Block calls and generates reorder tone
- Overwrite old CDR records with new CDR records
- Route calls to an attendant as non-CDR calls

CDR record formats

The system sends two types of records to the CDR output device, a date record and a call detail record.

Date record format

CDR sends date information to the CDR device once a day, at midnight, or when someone connects the device to the system. The record that the system generates at this time is a noncall record, and contains only the information shown in one of the date record formats.

Several date record formats exist:

- CDRU
- Printer
- TELESEER

The records sent to the TELESEER and the printer contain the date only. The records sent to the CDRU contain time. See the following tables for date record formats:

- [Table 18, Date record format to LSU, LSU-expand, unformatted, and customized](#), on page 384,
- [Table 19, Date record format for printer and expanded](#), on page 384
- [Table 20, Date record format for TELESEER 59 character, int-proc, int-direct, and int-ISDN](#), on page 384

Table 18: Date record format to LSU, LSU-expand, unformatted, and customized

Position	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 0 added if needed)
9	Slash (/)
10–11	Day (leading 0 added if needed)
12	Carriage return
13	Line feed
14–16	Null

Table 19: 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 20: 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

Customized CDR call record formats

You can use the customized record formats to define the call records for your system. You can determine the data elements that you want, and the position of the data elements in the record.

However, the device that you use to interpret the CDR data needs to be programmed to accept the data formats that you choose. Consult your Avaya representative before you use a customized record format.

Standard CDR call record formats

See the following tables for a description of the standard call record formats:

- [Table 21, CDR data format — ISDN TELESEER](#), on page 386
- [Table 22, CDR data format — ISDN TELESEER](#), on page 386
- [Table 23, CDR data format — enhanced TELESEER](#), on page 387
- [Table 24, CDR data format — 59 character](#), on page 388
- [Table 25, CDR data format — printer](#), on page 389
- [Table 26, CDR data format — ISDN printer](#), on page 390
- [Table 27, CDR data format — enhanced printer](#), on page 392
- [Table 28, CDR data format — LSU-expand](#), on page 393
- [Table 29, CDR data format — LSU](#), on page 394
- [Table 30, CDR data format — ISDN LSU](#), on page 395
- [Table 31, CDR data format — enhanced LSU](#), on page 396
- [Table 32, CDR data format — expanded](#), on page 397
- [Table 33, CDR data format — enhanced expanded](#), on page 399
- [Table 34, CDR data format — unformatted](#), on page 401
- [Table 35, CDR data format — enhanced unformatted](#), on page 402
- [Table 36, CDR data format — int process](#), on page 403
- [Table 37, CDR data format — int-direct](#), on page 404
- [Table 38, CDR data format — int-ISDN](#), on page 405

Table 21: CDR data format — ISDN TELESEER

Position	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	facilities restriction level (FRL)
55	inter-exchange carrier (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 22: CDR data format — ISDN TELESEER

Position	Description
1–3	Space
4–5	Time of day-hours
6–7	Time of day-minutes
1 of 2	

Table 22: CDR data format — ISDN TELESEER

Position	Description
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

2 of 2

Table 23: CDR data format — enhanced TELESEER

Position	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

1 of 2

Table 23: CDR data format — enhanced TELESEER

Position	Description
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

2 of 2

Table 24: CDR data format — 59 character

Position	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

1 of 2

Table 24: CDR data format — 59 character

Position	Description
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

2 of 2

Table 25: CDR data format — printer

Position	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

1 of 2

Table 25: CDR data format — printer

Position	Description
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
78–80	Outgoing circuit ID
81	Space
82	Feature flag
83	Carriage return
84	Line feed

2 of 2

Table 26: CDR data format — ISDN printer

Position	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

1 of 2

Table 26: CDR data format — ISDN printer

Position	Description
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
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

2 of 2

Table 27: CDR data format — enhanced printer

Position	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	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

1 of 2

Table 27: CDR data format — enhanced printer

Position	Description
83	Feature flag
84	Carriage return
85	Line feed

2 of 2

Table 28: CDR data format — LSU-expand

Position	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

1 of 2

Table 28: CDR data format — LSU-expand

Position	Description
60	Calling number (1st digit)
61	Space
62–63	Incoming circuit ID (tens, units)
64	Space
65	Feature flag
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

2 of 2

Table 29: CDR data format — LSU

Position	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 a 5-digit dial plan)
31–35	Account code (first 5 digits)

1 of 2

Table 29: CDR data format — LSU

Position	Description
36–42	Authorization code or digits 6–12 of the account code
43–44	Space or digits 13–14 of account code
45	FRL or digit 15 of the account code
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

2 of 2

Table 30: CDR data format — ISDN LSU

Position	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 a 5-digit dial plan)
31–35	Account code (digits 1–5)
36–42	Authorization code or digits 6–12 of the account code
43–44	INS or digits 13–14 of account code
45	INS (3rd digit), FRL, or digit 15 of the account code
46	Calling number (1st digit of a 5-digit calling number)

1 of 2

Table 30: CDR data format — ISDN LSU

Position	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

2 of 2

Table 31: CDR data format — enhanced LSU 1 of 2

Position	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 the account code
43–45	ISDN NSV
46	1st digit of a 5-digit calling number
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

1 of 2

Table 31: CDR data format — enhanced LSU 2 of 2

Position	Description
55	Carriage return
56	Line feed
57–59	Null

2 of 2

Table 32: CDR data format — expanded

Position	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
65	Space
66–72	Authorization code
73–76	Space
77	FRL
78	Space
79–81	Incoming circuit ID
82	Space

1 of 2

Table 32: CDR data format — expanded

Position	Description
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	Bearer capability class (BCC)
109	Space
110	Message-Associated User-to-User Signaling (MA-UII)
111	Space
112	Resource flag
113	Space
114–117	Packet count
118	Space
119	temporary-signaling connection (TSC) flag
120	Space
121–129	Reserved
130	Space
131	Carriage return
132	Line feed
133–135	Null

2 of 2

Table 33: CDR data format — enhanced expanded

Position	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
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

1 of 2

Table 33: CDR data format — enhanced expanded

Position	Description
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
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) / periodic pulse metering (PPM) count (1–5)
136–146	Reserved for future use
147	Carriage return
148	Line feed
149–151	Null

2 of 2

Table 34: CDR data format — unformatted

Position	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	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
101	Carriage return

1 of 2

Table 34: CDR data format — unformatted

Position	Description
102	Line feed
103–105	Null

2 of 2

Table 35: CDR data format — enhanced unformatted

Position	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
77–80	Incoming TAC
81–82	Node number
83–87	ISDN NSV
88–89	IXC code
90	BCC

1 of 2

Table 35: CDR data format — enhanced unformatted

Position	Description
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

2 of 2

Table 36: CDR data format — int process

Position	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
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)

1 of 2

Table 36: CDR data format — int process

Position	Description
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

2 of 2

Table 37: CDR data format — int-direct

Position	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

1 of 2

Table 37: CDR data format — int-direct

Position	Description
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
77–78	Outgoing circuit ID
79	Carriage return
80	Line feed

2 of 2

Table 38: CDR data format — int-ISDN

Position	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

1 of 3

Table 38: CDR data format — int-ISDN

Position	Description
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
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

2 of 3

Table 38: CDR data format — int-ISDN

Position	Description
95	Space
96–97	Node number
98	Space
99–101	INS
102	Space
103–106	IXC
107	Space
108	BCC
109	Space
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

3 of 3

Call detail record field descriptions

The following information describes the CDR data that the system collects for each call, and the length of each field. The information is right adjusted in the CDR record field, unless otherwise indicated. The customized field name appears, if the field name for customized records differ from the standardized records.

- **Access Code Dialed**

Customized field name: `code-dial`

Length: 3 or 4 digit

This field contains the access code that the user dials to place an outgoing call. The access can be the automatic route selection (ARS) access code, automatic alternate routing (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**

Custom field name: `code-used`

Length: 3 or 4 digits

This field is used only for outgoing calls when the system uses a trunk group that differs from the access code that the user dials. It is not used when a user dials a TAC. For example, your system might use an FAC for ARS. This field contains the access code of the trunk group that the system uses to route the call. When the access code that the user dials, and the access code that the systems uses are the same, this field is blank.

If you use ISDN or enhanced formats with TELESEER, LSU, or printer record types, this field contains the access code of the trunk group, even if the user dials the same access code.

- **Account Code**

Custom field name: `acct-code`

Length: 1 to 15 digits

This field is either blank, or it contains 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**

Custom field name: `attd-console`

Length: 2 digits

If an attendant participates in a call, this field contains the attendant console number of the attendant.

- **Authorization Code**

Custom field name: `auth-code`

Length: 4 to 13 digits

This field contains the authorization code that the user used to make the call. The system truncates an authorization code to 7 digits for formats other than customized formats.

Note that the authorization code for the non-ISDN format and the ISDN local storage init (LSU) format has fewer than 6 digits. The authorization code for the Enhanced LSU format has 5 digits. The system does not record the authorization on the 59-character record.

- **Bandwidth**

Length: 2 digits

This field contains the bandwidth of the wide band calls to support H0, H11, H12, and N x 64 kbps data rates. For Enhanced Expanded, Enhanced Unformatted, and customized record formats, this value in this field is the number of DSOs of 64-kbps channels that comprise a call.

- **Bearer Capability Class**

Custom field name: bcc

Length: 1 digit

This field contains the bearer capability class (BCC) for ISDN calls. The BCC identifies the type of ISDN call and distinguishes voice calls from different types of data. Any one of the following BCCs can 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 Link Access Procedure data (LAPD) protocol)

4 - Mode 0 (64 kbps data clear)

w - Wideband

For Intraswitch CDR records, the system provides information in the BCC field for wide band calls only.

- **Calling Number**

Custom field name: calling-num

Number of digits-standard: 1 to 10

Number of digits-custom: 1 to 15

For outgoing or intraswitch calls, this field contains the extension 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.

For formats in which the Calling Number field has 7 digits, the field contains the TAC of the incoming call.

For Unformatted records or Expanded records, this field contains the number of the calling party. If the number of the calling party is unavailable, this field is blank for both the Unformatted and the Expanded record formats.

For an outgoing, or an originating, NCA-TSC CDR record, this field contains the local extension of the noncall-associated/temporary-signaling connection (NCA-TSC) endpoint. This field is blank for terminating, tandem, or unsuccessful NCA-TSC CDR records.

- **Calling Number/Incoming TAC**

Custom field name: clg-num/in-tac

You can use this field on a customized record to display the calling number if the calling number is available.

If the calling party number is unavailable, this field contains the incoming TAC.

For outgoing calls, this field contains the calling extension.

- **Carriage Return**

Custom field name: `return`

The ASCII carriage return character, followed by a line feed, indicates the end of a call record.

- **Condition Code**

Length: 1 character

The condition code indicates the type of call that this record describes. For example, condition code C identifies a conference call, and condition code 7 identifies an ARS call.

[Table 39, Condition codes](#), on page 410 shows the condition codes for most record formats. The condition codes for the 59-character format differ from the condition codes of the other record types. The codes that apply to 59-character records appear in parentheses in the table.

Table 39: Condition codes

Condition codes	Description
0	Identifies an intraswitch call, which is 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 or a call with an extremely high message count TSC. An extremely long call is a call that lasts for 10 or more hours. An extremely high message count TSC is 9999 or more messages. When a call exceeds 10 hours, the system creates a call record with this condition code and a duration of 9 hours, 59 minutes, and 1–9 tenths of a minute. The system creates a similar call record with this condition code after each succeeding 10-hour period. When the call terminates, the system creates a final call record with a different condition code that identifies the type of call.
6 (E)	Identifies calls that the system does not record because system resources are unavailable. The CDR record includes the time and the duration of the outage. The system generates this condition code for: • Calls that the system routes to the attendant • Calls that require the CDR feature to overwrite records • ISDN calls that are not completed at the far end, if the Q.931 message includes the reason that the call was not completed. The system does not generate the condition code for ISDN calls that receive inband tones.
7 (G)	Identifies calls that use the AAR or ARS feature.
8 (H)	Identifies calls that are served on a delayed basis by the Ringback Queuing feature.

1 of 3

Table 39: Condition codes

Condition codes	Description
9 (I)	Identifies: • An incoming call • A tandem call • An incoming NCA-TSC call • A tandem NCA-TSC call
A	Identifies an outgoing call.
B	Identifies an adjunct-placed outgoing call.
C (L)	Identifies a conference call. For trunk CDR, the system creates a separate call record, with this condition code, for each incoming or outgoing trunk that is used during the conference call. If you disable ITCS and OTCS, the system records the extension of the originator of the conference call. The system does not record any other extension. If you disable ITCS, and you administer the originator of the conference call to use Intraswitch CDR, the system generates a call with this condition code whenever the originator of the conference dials a nontrunk conference participant. If ITCS is active, and you do <i>not</i> administer the originator of the conference call to use Intraswitch CDR, the system generates a call with this condition code whenever the originator of the conference dials an intraswitch conference participant.
E (N)	Identifies a call that the system does not complete because the following facilities to complete the call are unavailable: • Outgoing calls — The trunks are busy, and there is no queue exists — The trunks are busy, and the queue is full • Incoming calls — The extension is busy — The extension is unassigned This condition code also identifies an ISDN Call By Call Service Selection call that is unsuccessful because of an administered trunk usage allocation plan. Incoming trunk calls to a busy telephone do <i>not</i> generate a CDR record.
F	Identifies a call that the system does not complete because of one of the following conditions: • The originator of the calls has insufficient calling privileges • An NSF mismatch occurs for an ISDN call • An authorization mismatch occurs for a data call
G	Identifies a call that the system terminates to a ringing station.
2 of 3	

Table 39: Condition codes

Condition codes	Description
H	Notes that the system abandoned a ring call.
I	Identifies a call that the system terminates to a busy station.
J	Identifies an incoming trunk call that is a new connection that uses Additional Network Feature–Path Replacement (ANF-PR) or DCS with Rerouting. For more information on QSIG and ANF-PR, see the <i>Administrator's Guide for Avaya Communication Manager</i> .
K	Identifies an outgoing trunk call that is a new connection that uses ANF-PR or DCS with Rerouting. For more information on QSIG and ANF-PR, click here , or see the <i>Administrator's Guide for Avaya Communication Manager</i> .
M	Identifies an outgoing trunk calls that the system disconnects because the call exceeds the time allowed.
T	Identifies CDR records for call that meet the following conditions: • The Condition Code 'T' for Redirected Calls? field on the CDR System Parameters screen is set to y. • The incoming trunk group is DID. • The system automatically redirects an incoming call off of the server.

3 of 3

If the trunk-group CDR Reports field is set to **ring**, CDR records the ring time to answer or abandon for incoming calls that the trunk group originates. CDR also indicates if the incoming destination is busy. This record is separate from the normal call duration record that is printed for an answered call. This information is indicated by the condition code.

When a trunk group originates an incoming call with this option set that is terminated to an internal destination, the call is tracked from the time that ringing feedback is given to the originator. If the call is answered, a CDR record is printed with 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 condition code **H**, and the duration reflects the time between the start of ringing and the time that the call was abandoned. If the destination is busy, a CDR record is printed with 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 following matrix, [Table 40, Condition code override matrix](#), on page 413, defines the overrides.

To use this table, locate one of the condition codes that you want to compare in the row of condition codes at the top of the table. Then locate the other condition code that you want to compare in the column of condition code at the far left of the table. Then locate the intersection of these two condition codes. The condition code at the intersection of the two condition codes, is the condition code that the system uses. For example, assume that the system can assign condition 7 and condition code A to the same call. The system uses condition code 7 for the call.

If the intersection of the two condition codes contains a dash, the two conditions never occur for the same call.

Table 40: Condition code override matrix

Condition codes																
	0	1	4	6	7	8	9	A	B	C	E	F	J	K	M	T
0	—	0	4	6	0	—	—	—	B	C	—	—	—	—	—	—
1	0	—	4	6	1	—	9	1	B	C	E	—	J	K	—	—
4	4	4	—	6	4	4	4	4	4	4	—	—	J	K	—	—
6	6	6	6	—	6	6	6	6	6	6	6	6	6	6	—	—
7	0	1	4	6	—	7	9	7	B	C	E	F	J	K	—	—
8	—	—	4	6	7	—	—	8	B	C	E	—	—	K	—	—
9	—	9	4	6	9	—	—	—	—	C	E	F	—	—	—	—
A	—	1	4	6	7	8	—	—	B	C	E	F	—	—	—	—
B	B	B	4	6	B	B	—	B	—	B	E	F	—	K	—	—
C	C	C	4	6	C	C	C	C	B	—	—	—	J	K	M	—
E	—	E	—	6	E	E	E	E	E	—	—	—	E	E	—	—
F	—	—	—	6	F	—	F	F	F	—	—	—	F	F	—	—
J	—	J	J	6	J	—	—	—	—	J	E	F	—	—	—	—
K	—	K	K	6	K	K	—	—	—	K	E	F	—	—	—	—
M	—	—	—	—	—	—	—	—	—	M	—	—	—	—	—	—
T	—	—	—	—	—	—	—	—	—	—	—	—	—	—	—	—

- **Date**

You can include the date in customized CDR records only. The format of the date is based on the value of the CDR `Date Format` field on the **CDR System Parameters** screen.

- **Dialed Number**

Custom field name: `dialed-num`

Length: 23 digits

For an outgoing call, this field contains the number that the system user dials. For an incoming call, this field contains the extension of the system user that is called. If a Dialed Number Identification System (DNIS) exists, the field contains the implied extension for an incoming call. If the originator of the call dials more than eighteen digits, the system truncates the number or the extension to 18 digits. The system truncates the least significant digits, starting at the right. For example, if the originator of the call dials the number 1111111111111111852, the system truncates the digits 852.

— CDR Privacy

If CDR Privacy is active for the calling number, and this CDR record is for an outgoing call, the system modifies and records the number that the user dials according to the following procedure:

- The system first truncates the numbers that the user dials to 18 digits
- The system then replaces some of digits with blanks to ensure the privacy of the call.

— NCA-TSC or tandem NCA-TSC outgoing calls

For outgoing NCA-TSC or tandem NCA-TSC calls, this field contains the digits that the user dials, and that the system uses to establish a route to the far end server.

The field contains the local extension used as the NCA-TSC endpoint when the extension is for a terminating NCA-TSC. For an unsuccessful NCA-TSC outgoing call, this field is blank.

The field can contain a pound sign (#) or an *E* for either ARS calls or TAC calls when:

- A user dials a pound sign (#) at the end of a string of digits
- An outgoing call exceeds the interdigit-timeout interval that is in the ARS Analysis table
- 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, yields 1001E or 1001# in the **Dialed Number** field. The vector processing software uses the pound sign (#) or E to indicate the end of the string of digits that the user dials.

You administer the **CDR System Parameter** screen so the pound sign (#) or *E* need not be the last digit of the CDR record.

- **Duration**

Custom field name: `duration` or `sec-dur`

Length: 4 digits

This field contains the duration of the call, which the system records in hours (0 to 9), minutes (00 to 59), and tenths of minutes (0 to 9). The system rounds the duration of the call down in 6-second increments. For example, the system records the duration of a 5-second call as a duration of zero. If this field contains the value 9999, the call was in progress when a time change was made in the switch.

You can use the customized record format to report the duration of the call in hours, minutes, and seconds.

- **Feature Flag**

Custom field name: `feat - flag`

Length: 1 digit

The feature flag indicates whether a call received network answer supervision, and whether the call was interworked in the network. The duration of the call starts when the system receives the network answer.

You can administer the feature flag field on the CDR System Parameters screen to reflect whether the network reported an outgoing ISDN call as interworked.

- The number 0 in this field indicates either a voice call without network answer supervision or a call for which NCA-TSC is not established.
- The number 1 in this field indicates a data call without network answer supervision.

- The number 2 in this field indicates a voice call with network answer supervision, but interworked.
- The number 3 in this field indicates a data call with network answer supervision, but interworked.
- The number 4 in this field indicates a voice call with network answer supervision.
- The number 5 in this field indicates a data call with network answer supervision.

The time of the answer and the duration of the call are accurate if the feature flag indicates that the call received network answer supervision.

The time of the answer and the duration of the call might not be accurate, if the call does not receive network answer supervision, or the call receives answer supervision but is interworked with non-ISDN trunks.

Calls are considered data calls if the calls use a conversion resource, such as a modem, and the calls either originate or terminate on a data module.

- **Format Code**

Length: 2 digits

This field contains two values:

- 00 indicates no PPM.
- 03 indicates a PPM count in the digits record.

- **FRL**

Length: 1 digit

Facilities restriction levels (FRLs), numbered 0 through 7, are associated with the AAR and ARS features and define calling privileges. This field contains the following information if the call is an:

- Outgoing call, and an authorization code is not used to make the call, this field contains the FRL of the originating user FRL.
- Outgoing call, and an authorization code is used to make the call, this field contains the FRL that is associated with the authorization code that the user dials.
- Incoming or a tandem call, this field contains the FRL assigned to the incoming trunk group.
- Incoming tandem tie trunk call, this field contains either the FRL that is assigned to the tandem tie trunk or the TCM sent with the tandem tie trunk call. If an FRL was used to complete the call, the field contains the FRL. If a TCM was used to complete the call, the field contains the TCM. With ISDN calls, this field always contains the TCM, if the TCM was received.

You can administer CDR so this field contains disconnect information instead of the FRL. If you administer CDR so this field contains disconnect information for trunk CDR, the system records the information shown in [Table 41, Disconnect information for trunk CDR](#), on page 416.

Table 41: Disconnect information for trunk CDR

Data	Meaning
0	The system cannot determine which participant on the call dropped the call first.
1	The switch participant dropped the call first.
2	The central office (CO) participant dropped the call first.
3	Maintenance seized the trunk

For intraswitch CDR, the field contains the information shown in [Table 42, Intra-switch CDR call disconnect information](#), on page 416 instead of the FRL.

Table 42: Intra-switch CDR call disconnect information

Data	Meaning
0	The system cannot determine which participant on the call dropped the call first.
1	The calling participant dropped the call first.
2	The dialed participant dropped the call first.

- **Incoming Circuit Identification**

Custom field name: `in-crt-id`

Length: 3digits

This field contains the member number of a trunk within a trunk group that is 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 are inverted on the record. For example, the circuit ID 123 appears as 231 (tens, units, hundreds). If you want to change the order to hundreds, tens, units format, for example, 123, use the **Modified Circuit ID Display** field on the **CDR System Parameters** screen.

- **Incoming TAC**

Custom field name: `in-trk-code`

Length: 4 digits

This field contains the access code of the incoming trunk group.

- **INS**

Length: 3 digits

This field specifies the ISDN Network Service (INS) that is requested for a call. This field applies only to ISDN calls. Each network specific facility has a corresponding INS value, shown in [Table 43, Network-specific facility to INS mapping](#), on page 417.

This field also appears as ISDN NSV (network service value).

Table 43: Network-specific facility to INS mapping

Network specific facility	INS value
OUTWATS Band 0	33
OUTWATS Band 1–255	34–288
Network Operator	324
Pre subscribed 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 that the ISDN advice of charge function supplies.

- **ISDN NSV**

See INS.

- **IXC Code**

The length for 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 that is.

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 to 11 digit number. The IXC access numbers that are 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 central office (CO), the report contains a zero.

Length for ISDN formats: 3 or 4 digits

With an ISDN record format, this 3-digit or 4-digit field 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 digits or Enhanced). If a user dials a 7-digit IXC code, this field contains a zero.

- **Line Feed**

Length: 1 character

The ASCII line feed character comes after a carriage return to terminate CDR records.

- **MA-UII**

Length: 1 digit

Message Associated User-to-User Signaling shows the number of ISDN messages that contain user data that are sent on an outgoing call. This field contains a number from 0 to 9.

- **Node Number**

Custom field name: node-num

Length: 2 digits

This field identifies the DCS node number of a switch within a DCS arrangement. This number is the local ID, which is the same as the node number on the *Dial Plan* screen.

- **Null**

Length: 1 character

The Null character, usually in triplets, is used to terminate and divide CDR Records, when a receiving adjunct needs a record divider or terminator.

- **Outgoing Circuit Identification**

Custom field name: out-crt-id

Length: 3 digits

For outgoing calls, this field contains the member number of the trunk within a trunk group that is used on the call. 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 that is 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 are inverted on the record. For example, the circuit ID 123 appears as 231 (tens, units, hundreds). If you want to change the order to hundreds, tens, units format, for example, 123, use the **Modified Circuit ID Display** field on the *CDR System Parameters* screen.

- **Packet Count**

Custom field name: tsc_ct

Length: 4 digits

For ISDN TSCs, this field contains the number of ISDN-PRI USER INFO messages that are sent, received, or, for tandem TSCs, passed through the switch.

- **PPM**

Periodic Pulse Metering (PPM) contains the pulse counts that are transmitted over the trunk line from the serving CO. The pulse counts are used to determine call charges.

- **Resource Flag**

Custom field name: res-flag

Length: 1 digit

This field indicates whether the call was circuit switched or packet switched, whether a conversion resource was used, or if the call involved a multimedia application server interface (MASI) terminal or MASI trunk.

- 0 indicates circuit switched, no conversion device used
- 1 indicates packet switched, no conversion device used
- 2 indicates circuit switched, conversion device used
- 3 indicates packet switched, conversion device used
- 8 indicates a MASI call

- **Sec-dur**

For customized records only, you can use this field to set the duration field to record seconds instead of tenths of minutes.

- **Space**

Length: 0 to 40 characters

The ASCII space character separates other CDR fields or fills unused record locations.

- **TSC-Count**

Custom field name: `tsc_ct`

This field is the customized name for Packet Count. See Packet Count for more information.

- **TSC Flag**

Custom field name: `tsc_flag`

Length: 1 digit

This field describes call records that pertain to temporary signaling connections. When the value of this field is not equal to zero, this field indicates the status of the TSC. [Table 44, Encoding for the TSC flag](#), on page 419 shows the TSC flag encoding.

Table 44: Encoding for the TSC flag

Code	Description
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 by SETUP, no congestion control (applicable to terminating node).
5	Call Associated TSC received and accepted by 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).

1 of 2

Table 44: Encoding for the TSC flag

Code	Description
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 (NCA) TSC received, accepted, no congestion control (applicable to terminating node). NCA TSC received, accepted, no congestion control (applicable to terminating node).
B	NCA TSC requested, accepted, congestion control (applicable to originating node). NCA TSC received, accepted, congestion control (applicable to terminating node).
C	NCA 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.

2 of 2

- **Time**

This field contains the time that the call ended, or Call Splitting is active, the time that a user dropped from a multiparty call.

- **VDN**

Custom field name: vdn

Length: 5 digits

This field is available on customized records only. The call record contains the VDN extension number. If VDN Return Destination is active, this field contains the first VDN that the caller accessed.

Hardware requirements for Call Detail Recording

The Call Detail Recording feature requires the following hardware:

- None

Administering Call Detail Recording

The following steps are part of the administration process for the Call Detail Recording feature:

- [Assigning Forced Entry of Account Codes](#)
- [Assigning privacy digits for a user](#)
- [Administering the CDR system parameters](#)
- [Administering CDR for a trunk group](#)
- [Administering CDR for a data module](#)
- [Identifying the Inter Exchange Carrier](#)
- [Administering CDR for the paging ports](#)
- [Administering the Intra-Switch CDR](#)

This section describes:

- Any prerequisites for administering the Call Detail Recording feature
- The screens that you use to administer the Call Detail Recording feature
- Complete administration procedures for the Call Detail Recording feature

Prerequisites for administering Call Detail Recording

You must complete the following actions before you can administer the Call Detail Recording (CDR) feature:

- Ensure that the CDR account code is available on your system.

To ensure that the CDR account code is available on your system:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the *Feature Access Codes (FAC)* screen ([Figure 86, Feature Access Code \(FAC\) screen](#), on page 422).

Figure 86: Feature Access Code (FAC) screen

FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1	Access Code:	*00
Abbreviated Dialing List2	Access Code:	*01
Abbreviated Dialing List3	Access Code:	*02
Abbreviated Dial - Prgm Group List	Access Code:	*03
Announcement	Access Code:	*04
Answer Back	Access Code:	*05
Auto Alternate Routing (AAR)	Access Code:	8
Auto Route Selection (ARS) -	Access Code 1:	*9
Automatic Callback Activation:	Access Code 2:	9
Call Forwarding Activation Busy/DA:	Deactivation:	*06 #06
Call Park	Deactivation:	*07 All: *08 #09
Call Pickup	Access Code:	*10
CAS Remote Hold/Answer Hold-Unhold	Access Code:	*11
CDR Account Code	Access Code:	*50
Change COR	Access Code:	*12
Change Coverage	Access Code:	*13
Contact Closure	Open Code:	
Contact Closure	Pulse Code:	
	Close Code:	

- 2 In the CDR Account Code Access Code field, type the code that you want a user to enter before the user enters a CDR account code number.

For more information, see the “Feature Access Code” feature.

- 3 Press **Enter** to save your changes.

- Ensure that call classification is enabled for your system, if you want to use call classification.

View the **Optional Features** screen, and ensure that the Answer Supervision by Call Classifier field is set to **y**. If the Answer Supervision by Call Classifier field is set to **n**, your system is not enabled for call classification. Contact your Avaya representative if you want to use call classification.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

Screens for administering Call Detail Recording

Screen name	Purpose	Fields
<i>CDR System Parameters</i>	Define CDR for the system	All
<i>Class of Restriction</i>	Add Forced Entry of Account Codes (FEACs) to a Class of Restriction (COR).	Forced Entry of Account Codes
<i>Data Module</i>	Define a data module that uses CDR.	All
<i>Feature Access Code (FAC)</i>	Assign the FAC that a user to enters before the user enters an account code for CDR.	CDR Account Code Access Code
<i>Inter-exchange Carrier Codes</i>	Identify the inter-exchange carrier (IXC) in the CDR record.	All
<i>Intraswitch CDR</i>	Provide information so CDR records intraserver calls.	All
<i>Loudspeaker Paging and Code Calling Access</i>	Specify CDR record collecting for the paging ports.	CDR
<i>Optional Features</i>	Enable FEAC.	Forced Entry of Account Codes
<i>Station</i>	Assign a COR for FEAC to a user.	COR
<i>Trunk Groups</i>	Assign CDR reports and timers to a trunk group.	• CDR Reports • Answer Supervision Timeout • Disconnect Supervision
	Assign a COR for FEAC to a trunk group.	COR

Assigning Forced Entry of Account Codes

Prerequisites

You must complete the following actions before you can assign FEAC for all calls:

- View the **Optional Features** screen, and ensure that the Forced Entry of Account Codes field is set to **y**. If the Forced Entry of Account Codes field is set to **n**, your system is not enabled for FEAC for all calls. Contact your Avaya representative if you want to use FEAC for all calls

To view the **Optional Features** screen, type **display System-Parameters Customer-Options**. Press **Enter**.

- Assign FEAC for all calls.

1 Type **change system-parameters cdr**. Press **Enter**.

The system displays the **CDR System Parameters** screen ([Figure 87, CDR System Parameters](#), on page 424).

Figure 87: CDR System Parameters

change system-parameters cdr		Page 1 of 2
CDR SYSTEM PARAMETERS		
Node Number (Local PBX ID): 18	CDR Date Format: month/day	
Primary Output Format: customized	Primary Output Endpoint: CDR1	
Secondary Output Format:		
Use ISDN Layouts? n		
Use Enhanced Formats? n	Condition Code 'T' For Redirected Calls? n	
Modified Circuit ID Display? n	Remove # From Called Number? y	
Record Outgoing Calls Only? n	Intra-switch CDR? y	
Suppress CDR for Ineffective Call Attempts? y	Outg Trk Call Splitting? y	
Disconnect Information in Place of FRL? n	Outg Attd Call Record? y	
	Interworking Feat-flag? y	
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 Agent ID on Incoming? n	Record Agent ID on Outgoing? y	
Inc Trk Call Splitting? y	Inc Attd Call Record? n	
Record Non-Call-Assoc TSC? n	Call Record Handling Option: warning	
Record Call-Assoc TSC? n	Digits to Record for Outgoing Calls: outpulsed	
Privacy - Digits to Hide: 0	CDR Account Code Length: 4	

2 In the Force Entry of Acct Code for Calls Marked on Toll Analysis Form? field, perform one of the following actions:

- Type **y** if you want the system to require that a user enter an account code for all calls.
- Type **n** if you do not want the system to require that a user enter an account code for all calls.

The information in the Force Entry of Acct Code for Calls Marked on Toll Analysis Form? field does not override other call restrictions that the user might have.

- Assign FEAC to a COR.

1 Type **change cor *n***, where *n* is the number of the COR to which you want to assign FEAC. Press **Enter**.

The system displays the **Class of Restriction** screen ([Figure 88, Class of Restriction screen](#), on page 425).

Figure 88: Class of Restriction screen

change cor 1	CLASS OF RESTRICTION	Page 1 of 4
COR Number: 1 COR Description:		
FRL: 7 APLT? y		
Can Be Service Observed? y	Calling Party Restriction: none	
Can Be A Service Observer? y	Called Party Restriction: none	
Partitioned Group Number: 1	Forced Entry of Account Codes? y	
Priority Queuing? n	Direct Agent Calling? y	
Restriction Override: none	Facility Access Trunk Test? n	
Restricted Call List? n	Can Change Coverage? n	
Access to MCT? y Fully Restricted Service? n		
Group II Category For MFC: 7	Hear VDN of Origin Annc.? y	
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 Phone)? y	
	Can Be Picked Up By Directed Call Pickup? n	
	Can Use Directed Call Pickup? n	
	Group Controlled Restriction: inactive	

2 In the Forced Entry of Account Codes field, perform one of the following actions:

- Type **y** if you want all users to enter an account code for calls that require an account code.
- Type **n** if you do not want all users to enter an account code for calls that require an account code.

3 Press **Enter** to save your changes.

To assign FEAC:

1 Type **change station *n***, where *n* is the number of the user extension to which you want to assign FEAC. Press **Enter**.

The system displays the **Station** screen, ([Figure 89, Station screen](#), on page 426).

Figure 89: Station screen

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?	
Model:		
Survivable GK Node Name:	Media Complex Ext:	
	IP Softphone? y	
	Remote Office Phone? y	

- 2 In the COR field, type the number of the COR that requires the user to enter an account code.
- 3 Press **Enter** to save your changes.
- 4 Type **change trunk-group n**, where *n* is the number of the trunk group to which you want to assign FEAC. Press **Enter**.

The system displays the *Trunk Group* screen ([Figure 90, Trunk Group screen](#), on page 426).

Figure 90: Trunk Group screen

change trunk-group 1		Page 1 of 22
TRUNK GROUP		
Group Number: 1	Group Type: isdn	CDR Reports: y
Group Name: GERT tg1 tf Lulu tg21 PRI	COR: 1	TN: 1 TAC: 354
Direction: two-way	Outgoing Display? y	Carrier Medium: PRI/BRI
Dial Access? y	Busy Threshold: 255	Night Service:
Queue Length: 0		
Service Type: tie	Auth Code? n	TestCall ITC: rest
	Far End Test Line No:	
TestCall BCC: 4		
TRUNK PARAMETERS		
Codeset to Send Display: 6	Codeset to Send National IEs: 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? n	
Calling Number - Delete:	Insert:	Digital Loss Group: 13
Bit Rate: 1200	Synchronization: async	Numbering Format: pub-unk
Disconnect Supervision - In? y Out? n	Duplex: full	
Answer Supervision Timeout: 0		

- 5 In the COR field, type the number of the COR that requires a user to enter an account code for calls that use this trunk group.
- 6 Press **Enter** to save your changes.

Assigning privacy digits for a user

Prerequisites

You must complete the following actions before you can assign privacy digits for a user:

- Assign the system-wide privacy digits
 - 1 Type **change system-parameters cdr**. Press **Enter**.
The system displays the **CDR System Parameters** screen ([Figure 91, CDR System Parameters screen](#), on page 427).

Figure 91: CDR System Parameters screen

change system-parameters cdr		Page 1 of 2
CDR SYSTEM PARAMETERS		
Node Number (Local PBX ID): 18	CDR Date Format: month/day	
Primary Output Format: customized	Primary Output Endpoint: CDR1	
Secondary Output Format:		
Use ISDN Layouts? n	Condition Code 'T' For Redirected Calls? n	
Use Enhanced Formats? n	Remove # From Called Number? y	
Modified Circuit ID Display? n	Intra-switch CDR? y	
Record Outgoing Calls Only? n	Outg Trk Call Splitting? y	
Suppress CDR for Ineffective Call Attempts? y	Outg Attd Call Record? y	
Disconnect Information in Place of FRL? n	Interworking Feat-flag? y	
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 Agent ID on Outgoing? y	
Record Agent ID on Incoming? n	Inc Attd Call Record? n	
Inc Trk Call Splitting? y	Call Record Handling Option: warning	
Record Non-Call-Assoc TSC? n	Digits to Record for Outgoing Calls: outputted	
Record Call-Assoc TSC? n	CDR Account Code Length: 4	
Privacy - Digits to Hide: 0		

- 2 In the Privacy - Digits to Hide field, type the number of user-dialed digits that the system replaces with blanks when the system generates a CDR record of the call.
- 3 Press **Enter** to save your changes.

To assign CDR privacy to a user:

- 1 Type **change station n**, where **n** is the number of the user extension to which you want to assign CDR privacy. Press **Enter**.
The system displays the **Station** screen ([Figure 92, Station screen](#), on page 428).

Figure 92: Station screen

change station 75001	STATION	Page 2 of X
FEATURE OPTIONS		
LWC Reception? 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: _____	Select Last Used Appearance? n	
AUDIX Name: _____	Coverage After Forwarding? _	
Recall Rotary Digit? n	Multimedia Early Answer? n	
Remote Softphone Emergency Calls: as-on-local	Direct IP-IP Audio Connections? n	
Emergency Location Ext: 75001	IP Audio Hairpinning? n	

- 2 Page through the screens until you see the CDR Privacy? field.
- 3 In the CDR Privacy? field, perform one of the following actions:
 - Type **y** if you want the system to replace some of the digits that the user dials, when the system generates a CDR record of the call.
 - Type **n** if you do not want the system to replace some of the digits that the user dials, when the system generates a CDR record of the call.
- 4 Press **Enter** to save your changes.

Administering the CDR system parameters

To administer the CDR system parameters for your system:

- 1 Type **change system-parameters cdr**. Press **Enter**.

The system displays the *CDR System Parameters* screen ([Figure 93, CDR System Parameters screen](#), on page 429).

Figure 93: CDR System Parameters screen

change system-parameters cdr		Page 1 of 2
CDR SYSTEM PARAMETERS		
Node Number (Local PBX ID): 18	CDR Date Format: month/day	
Primary Output Format: customized	Primary Output Endpoint: CDR1	
Secondary Output Format:		
Use ISDN Layouts? n	Condition Code 'T' For Redirected Calls? n	
Use Enhanced Formats? n	Remove # From Called Number? y	
Modified Circuit ID Display? n	Record Outgoing Calls Only? n	
Record Outgoing Calls Only? n	Intra-switch CDR? y	
Suppress CDR for Ineffective Call Attempts? y	Outg Trk Call Splitting? y	
Disconnect Information in Place of FRL? n	Outg Attd Call Record? y	
	Interworking Feat-flag? y	
Force Entry of Acct Code for Calls Marked on Toll Analysis Form? y	Calls to Hunt Group - Record: member-ext	
Record Called Vector Directory Number Instead of Group or Member? n	Record Agent ID on Incoming? n	
Record Agent ID on Incoming? n	Record Agent ID on Outgoing? y	
Inc Trk Call Splitting? y	Inc Attd Call Record? n	
Record Non-Call-Assoc TSC? n	Call Record Handling Option: warning	
Record Call-Assoc TSC? n	Digits to Record for Outgoing Calls: outpulsed	
Privacy - Digits to Hide: 0	CDR Account Code Length: 4	

- 2 In the Inc Trk Call Splitting? field, perform one of the following actions:
 - Type **y** if you want an incoming trunk call, that is conferenced or transferred to a local extension that is optioned for Intraswitch CDR, to produce an incoming trunk call record.
If you set the Inc Trk Call Splitting? field to y, you must set the Record Outgoing Calls Only field to **n**.
 - Type **n** if you do not want an incoming trunk call, that is conferenced or transferred to a local extension that is optioned for Intraswitch CDR, to produce an incoming trunk call record.
- 3 In the Record Outgoing Calls Only field, perform one of the following actions:
 - Type **y** if you only want the system to record outgoing calls.
 - Type **n** if you want the system to record both incoming and outgoing calls.
- 4 In the Call Record Handling Option field, perform one of the following actions:
 - Type **reorder** if you want the system block calls generate reorder tone when the buffer is full. If you choose this option, no one can make or receive calls if the system cannot generate CDR records for the calls.
 - Type **warning** if you want the system stop recording call when the buffer is full. If the buffer is full, and you choose this option, the system generates a minor alarm.
Warning is the system default for this field. Note that if you change the default, the system might redirect the ACD calls and the vector calls the system records for CDR.
 - Type **attendant** if you want the system to route all the calls to the attendant as non-CDR calls.

Note that if you change the system default of **warning**, ACD calls and vector calls that are measured by CDR might be redirected.

Note that the system displays the **Call Record Handling Option** field only for DEFINITY R. The system uses the information in this field to control call routing when:

- New calls come in.
- The CDR link is not operating.
- The buffer is full.

- 5 In the **Calls to Hunt Group-Record** field, perform one of the following actions:
 - Type **member-ext** if you want the system to record the extension of the telephone or data terminal where the call terminates.
 - Type **group-ext** if you want the system to record the extension that the user dials.
- 6 In the **CDR Account Code Length** field, type the number of digits that you want the system to record when a user enters an account code. For some record formats, the system overwrites the information in other fields if the account code is too long.
- 7 In the **CDR Date Format** field, perform one of the following actions:
 - Type **month/day** if you want to use the month and day date format for the date stamp that starts each new day of call records.
 - Type **day/month** if you want to use the day and month date format for the date stamp that starts each new day of call records.
- 8 In the **Condition Code 'T' for Redirected Calls** field, perform one of the following actions:
 - Type **y** if you want the system to record condition code *T* for both CDR records of the call that the system automatically redirects off the server that runs Communication Manager.
 - Type **n** if you want the system to record the condition codes that are usually associated with the **Record Outgoing Call Only** field for calls that the system automatically redirects off the server that runs Communication Manager.
- 9 In the **Digits to Record for Outgoing Calls** field, perform one of the following actions:
 - Type **dialed** to record the digits that a user dials.
 - Type **outpulsed** to record the digits that the software actually sends out over the trunk. This information includes any additions or deletions that take place during routing.
- 10 In the **Disconnect Information in Place of FRL** field, perform one of the following actions:
 - Type **y** if you want the system to replace the **Facility Restriction Level (FRL)** field with the call disconnect information. You can use the call disconnect information to isolate problems between the DEFINITY R and the telephone network.
 - Type **n** if you want the system to record the facilities restriction level (FRL) of the call.
- 11 In the **EIA Device Bit Rate** field, type the baud of the CDR device that is connected to the Electronic Industries Association (EIA) port. The valid bauds for this field are:
 - **300**
 - **1200**
 - **2400**
 - **4800**
 - **9600**

Note that the system displays this field only if either the **Primary Output Format** field or the **Secondary Output Format** field is set to **cia**, and then only for a **DEFINITY SI**.

- 12** In the **Inc Attd Call Record** field, perform one of the following actions:

- Type **y** if you want the system to generate separate records of the attendant portions of incoming calls that the attendant transfers or conferences.
- Type **n** if you do not want the system to generate separate records of the attendant portions of incoming calls that the attendant transfers or conferences.

Note that the system displays this field only when the **Inc Trk Call Splitting** field is set to **y**.

- 13** In the **Inc Trk Call Splitting** field, perform one of the following actions:

- Type **y** if you want the system to create separate records for each portion of an incoming call that is transferred or conferenced.
- Type **n** if you do not want the system to create separate records for each portion of an incoming call that is transferred or conferenced.

Note that the system only displays this field when the **Record Outgoing Calls Only** field on the **System Parameters CDR** screen is set to **n**.

- 14** In the **Interworking Feat-flag** field, perform one of the following actions:

- Type **y** if you want the system to record, in the **Feature Flag** field of a CDR record, that a call is an interworked outgoing ISDN call.

An interworked call is a call that passes through more than one ISDN node.

- Type **n** if you want the system to record, in the **Feature Flag** field of a CDR record, that there is no answer supervision for interworked calls.

- 15** In the **Intra-Switch CDR** field, perform one of the following actions:

- Type **y** if you want the system to record calls within the server. If you type **y**, you must administer the **Intraswitch CDR** screen to specify the extensions that you want the system to monitor.
- Type **n** if you do not want the system to record calls within the server.

- 16** In the **Modified Circuit ID Display** field, perform one of the following actions:

- Type **y** if you want the system to display the circuit ID in the actual format of 100's, 10's units, for example, if you want the system to display circuit ID 123 as 123. Verify that the output device of your system can accept this format.
- Type **n** to display the circuit ID in its default format (10's, units, 100's). For example, the system displays circuit ID 123 appears as 231.

The information in the **Modified Circuit ID Display** field pertains to the following CDR output formats:

- Printer
- TELESEER
- 59-character

The **In the Node Number (Local PBX ID)** field is a display-only field that is set to the distributed communications system (DCS) switch node number in a network of switches.

17 In the **Outg Attd Call Record** field, perform one of the following actions:

- Type **y** if you want the system to generate separate records of the attendant portions of outgoing calls that the attendant transfers or conferences.
- Type **n** if you do not want the system to generate separate records of the attendant portions of outgoing calls that the attendant transfers or conferences.

Note that the system displays this field only when the **Outg Trk Call Splitting** field is set to **y**.

18 In the **Outg Trk Call Splitting** field, perform one of the following actions:

- Type **y** if you want the system to create separate records for each portion of an outgoing call that is transferred or conferenced.
- Type **n** if you do not want the system to create separate records for each portion of an outgoing call that is transferred or conferenced.

19 In the **Primary Output Endpoint** field, perform one of the following actions:

- Type **eia** if the system uses the EIA port to connect the CDR device. This option is not valid for DEFINITY R systems.
- Type the extension of the data module that links the primary output device to the server.
- Type **CDR1** if the CDR device connects over a TCP/IP link, and the TCP/IP link is defined as CDR1 on the **IP Services** screen.
- Type **CDR2** if the CDR device connects over a TCP/IP link, and the TCP/IP link is defined as CDR2 on the **IP Services** screen.

20 In the **Primary Output Format** field, perform one of the following actions:

- Type **customized** if you do not want to use the standard CDR record formats. If you use a customized record format, your system must have call accounting software that is also customized to receive the customized records. Talk with your call accounting vendor before you select this option.
- Type **printer** if you want the system to send the CDR record formats to a printer instead of to a record collection system or to a call accounting system.
- Type the standard record format that you want to use on your system. The valid standard record formats are:

- **59-char**
- **expanded**
- **lsu**
- **lsu-expand**
- **int-direct**
- **int-isdn**
- **int-process**
- **TELESEER**
- **unformatted**

The standard record format that you choose must be compatible with the call accounting software on your system. To ensure that the standard record format that you choose is compatible with your call accounting system, talk with your vendor or see the call accounting system documentation.

- 21** In the **Privacy - Digits to Hide** field, type the number of dialed number digits that you want the system to hide for an extension with the **CDR Privacy** field on the **Station** record set to **y**. The valid entries for the **Privacy - Digits to Hide** field are **0** through **7**.

The system hides the dialed digits from right to left. If you type **4** in the **Privacy - Digits to Hide** field, and the user dials 5551234, the system records the dialed number as 555.

- 22** In the **Record Agent ID on Incoming** field, perform one of the following actions:
- Type **y** if you want the system to record the login ID of the EAS agent in the **Dialed Number** field of the CDR record.
 - Type **n** if you want the system to record the physical extension in the **Dialed Number** field of the CDR record.

The system displays the **Record Agent ID on Incoming** field only if the **Expert Agent Selection (EAS)** field on the **Optional Features** screen is set to **y**.

You cannot use both the **Called VDN** field and the **Agent Login ID Instead of Group or Member** field. Only one of these fields can be set to **y**.

- 23** In the **Record Agent ID on Outgoing** field, perform one of the following actions:
- Type **y** if you want the system to record the login ID of the EAS agent in the **Dialed Number** field of the CDR record.
 - Type **n** if you want the system to record the physical extension in the **Dialed Number** field of the CDR record.

The system displays the **Record Agent ID on Outgoing** field only if the **Expert Agent Selection (EAS)** field on the **Optional Features** screen is set to **y**.

- 24** In the **Record Call-Assoc TSC** field, perform one of the following actions:
- Type **y** if you want the system to generate records for call-associated temporary signaling connections.
- Consider the capacity of your call collection device before you decide to generate records for call-associated noncall-associated/temporary-signaling connection (TSCs).
- Type **n** if you do not want the system to generate records for call-associated TSCs.

- 25** In the **Record Called Vector Directory Number Instead of Group or Member** field, perform one of the following actions:
- Type **y** if you want the system to record the Vector Directory Number (VDN) in the **Dialed Number** field of the CDR record for calls that the system routes to a hunt group because of a vector. If the system routes a call through more than one VDN, the system records the first VDN in the **Dialed Number** field of the CDR record.
 - Type **n** if you want the system to record the group number or the member number in the **Dialed Number** field of the CDR record for calls that the system routes to a hunt group because of a vector.

You cannot use both the **Called VDN** field and the **Agent Login ID Instead of Group or Member** field. Only one of these fields can be set to **y**.

- 26** In the **Record Non-Call-Assoc TSC** field, perform one of the following actions:
- Type **y** if you want the system to create records for noncall-associated temporary signaling connections.

Consider the capacity of your call collection device before you decide to generate records for call-associated TSCs.

- Type **n** if you do not want the system to generate records for noncall-associated TSCs.

A TSC is a virtual connection that is established within an ISDN D-channel. For more information, see the *DEFINITY® Communications System Generic 2.2 and Generic 3 V2 DS1/CEPT1/ISDN PRI Reference*.

- 27** In the **Record Outgoing Calls Only** field, perform one of the following actions:

- Type **y** if you want the system to record only outgoing calls.
- Type **n** if you want the system to record incoming and outgoing calls.

- 28** In the **Remove # From Called Number** field, perform one of the following actions:

- Type **y** if you want the system remove the pound sign (#) or the letter E from the **Dialed Number** field of the call detail record.

Verify that your output device can accept this format.

- Type **n** if you want the system to record the trailing pound sign (#) or the letter E in the **Dialed Number** field whenever interdigit time out occurs or users dial # (pound sign) to indicate the end of a dialed string.

- 29** In the **Secondary Output Endpoint** field, perform one of the following actions:

- Type **eia** if the system uses the EIA port to connect the CDR device. This option is not valid for DEFINITY R systems.
- Type the extension of the data module that links the primary output device to the server.
- Type **CDR1** if the CDR device connects over a TCP/IP link, and the TCP/IP link is defined as CDR1 on the **IP Services** screen.

The system displays the **Secondary Output Endpoint** field when the you administer the **Secondary Output Format** field.

- 30** In the **Secondary Output Format** field, type the formats that you want your system to use for a secondary output device. The valid formats for a secondary out device are:

- **customized**
- **int-direct**
- **int-process**
- **lsu**
- **unformatted**



CAUTION:

Only qualified service personnel should administer a secondary output device. This option can cause loss of data when the buffer contains large amounts of data.

31 In the **Suppress CDR for Ineffective Call Attempts** field, perform one of the following actions:

- Type **y** if you want the system to ignore ineffective call attempts. Perform this action if you have limited storage space for CDR records and the CDR records often overrun the buffer.
- Type **n** if you want the system to record ineffective call attempts.

Ineffective call attempt information shows you how often your users cannot place outgoing calls, or if numerous incoming calls are not completed. You can also use the information to document attempts to contact a client when you use ISDN trunks.

Your system requires more space for records if the system records ineffective call attempts than if the system does not record ineffective call attempts.

Ineffective call attempts are calls that the system blocks because:

- The user does not have sufficient calling privileges
- All the outgoing trunks are busy
- Incoming or outgoing trunks are unavailable because of trunk usage allocation for ISDN Call-by-Call Service Selection trunks
- Incoming calls have an network-specific facility (NSF) mismatch
- A cause value is provided for ISDN calls that are not complete at the far end

The system record the ineffective call attempt as condition code E.

32 In the **Use Enhanced Formats** field, perform one of the following actions:

- Type **y** if you want to use the enhanced version of the specified primary output format in your system. You cannot use enhanced formats and ISDN formats at the same time.
- Type **n** if you do not want to use the enhanced version of the specified primary output format.

The enhanced formats provide additional information about the time a call is in a queue and about ISDN call charges. The **Use Enhanced Formats** field pertains to following output formats:

- Expanded
- TELESEER
- Lsu
- Printer
- Unformatted

33 In the **Use ISDN Layouts** field, perform one of the following actions:

- Type **y** to use the ISDN version of the specified primary output format. You cannot use ISDN formats and enhanced formats at the same time.
- Type **y** if you do not want to use the ISDN version of the specified primary output format.

The ISDN formats provide more accurate information about the IXC and the ISDN network services that are used for a call. The **Use ISDN Layouts** field pertains to the following output formats:

- Lsu
- Printer
- Any format with an ISDN layout, such as TELESEER

Field descriptions for page 2

Figure 94: CDR System Parameters screen

change system-parameters cdr		Page 2 of 2	
CDR SYSTEM PARAMETERS			
Data Item - Length	Data Item - Length	Data Item - Length	
1: date - 6	17: clg-num/in-tac - 10	33: -	
2: space - 1	18: space - 1	34: -	
3: time - 4	19: bandwidth - 2	35: -	
4: space - 1	20: space - 1	36: -	
5: sec-dur - 5	21: in-crt-id - 3	37: -	
6: space - 1	22: space - 1	38: -	
7: cond-code - 1	23: out-crt-id - 3	39: -	
8: space - 1	24: space - 1	40: -	
9: feat-flag - 1	25: return - 1	41: -	
10: space - 1	26: line-feed - 1	42: -	
11: code-dial - 4	27: -	43: -	
12: space - 1	28: -	44: -	
13: code-used - 4	29: -	45: -	
14: space - 1	30: -	46: -	
15: dialed-num - 10	31: -	47: -	
16: space - 1	32: -	48: -	
Record length = 67			

The system displays this second **CDR System Parameters** screen only if the Primary Record Format is set to **customized**.

- 34** In the **Data Item** field, type the data items in the order that you want the items to appear in the customized CDR record.

You must include at least three fields on this **CDR System Parameters** screen if you want to have a customized CDR record. The first field can be any field that you choose from [Table 45, Valid data item entries](#), on page 436. The last two fields items in a record must be **line-feed** and **return**, in that order.

Table 45: 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
bandwidth	2	ma-uui	1
bcc	1	node-num	2
calling-num	15	null	1
clg-pty-cat	2	out-crt-id	3

1 of 2

Table 45: Valid data item entries

Data item	Length	Data item	Length
clg-num/in-tac	10	ppm	5
code-dial	4	res-flag	1
code-used	4	return	1
cond-code	1	sec-dur	5
date	6	space	1
dialed-num	23	time	4
duration	4	tsc_ct	4
feat-flag	1	tsc_flag	1
fri	1	vdn	1
in-crt-id	3		
in-trk-code	4		
line-feed	1		
2 of 2			

35 In the **Length** field, type the maximum length of each data item, if the length of the data item differs from the default length.

You must type **6** for the length of any date field to ensure proper output.

In some cases, the system enforces a default field length.

The **Record Length** field is a display-only field that contains the sum of all the numbers that you type in all the of the **Length** fields. If you change a **Length** field, the system automatically changes the number in the **Record Length** field.

36 Press **Enter** to save your changes.

Administering CDR for a trunk group

To administer CDR for a trunk group:

- 1** Type **change trunk-group *n***, where *n* is the number of the trunk group for which you want to administer CDR. Press **Enter**.

The system displays the **Trunk Group** screen ([Figure 95, Trunk Group screen](#), on page 438).

Figure 95: Trunk Group screen

change trunk-group 1		Page 1 of 22
TRUNK GROUP		
Group Number: 1	Group Type: isdn	CDR Reports: y
Group Name: GERT tg1 tf Lulu tg21 PRI	COR: 1	TN: 1 TAC: 354
Direction: two-way	Outgoing Display? y	Carrier Medium: PRI/BRI
Dial Access? y	Busy Threshold: 255	Night Service:
Queue Length: 0		
Service Type: tie	Auth Code? n	TestCall ITC: rest
	Far End Test Line No:	
TestCall BCC: 4		
TRUNK PARAMETERS		
Codeset to Send Display: 6	Codeset to Send National IEs: 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? n	
	Digital Loss Group: 13	
Calling Number - Delete:	Insert:	Numbering Format: pub-unk
Bit Rate: 1200	Synchronization: async	Duplex: full
Disconnect Supervision - In? y Out? n		
Answer Supervision Timeout: 0		

2 In the **Answer Supervision Timeout** field, perform one of the following actions:

- If the **Receive Answer Supervision** field is set to **n**, type the number of seconds that you want the system to wait, before the system acts as if answer supervision was received from the far end.

The system uses the timing information for outgoing and two-way trunks. For a cut-through operation, the system starts tracking the time after the system sends each outgoing digit. The system stops tracking the time after the far end sends answer supervision. If the timer expires, the system acts as if answer supervision was received. With a senderized operation, the system starts tracking the time after the system sends the last collected digit.

- If the **Receive Answer Supervision** field is set to **y**, type **0**.

Note that the **Answer Supervision Timeout** field does not override answer supervision sent from the network or from DS1 port circuit timers. To control answer supervision that is sent by DS1 firmware, you must set the **Outgoing End of Dial (sec)** field on the **Administrable Timers** page of the **Trunk Group** screen.

3 In the **CDR Reports** field, perform one of the following actions:

- Type **y** if you want the system to generate records for all the outgoing calls on this trunk group.

If the **Record Outgoing Calls Only** field on the **CDR System Parameters** screen is **n**, then incoming calls on this trunk group also generate call detail records.

- Type **n** if you do not want calls that use this trunk group to generate CDR records.

- Type **r**, for a ring interval, if you want the system to generate both incoming and outgoing CDR records. The system also generates the ring interval CDR records that [Table 46, CDR ring interval records](#), on page 439 shows.

Table 46: CDR ring interval records

Type	Description
Abandoned calls	The system creates a record with condition code <i>H</i> . Condition code <i>H</i> indicates the interval from the start of ringing until the call was abandoned.
Answered calls	The system creates a record with condition code <i>G</i> . Condition code <i>G</i> indicates the interval from the start of ringing until the call was answered.
Calls to busy stations	The system creates a record with condition code <i>I</i> . condition code <i>I</i> indicates a recorded interval of 0.

For ISDN trunk groups, the **Charge Advice** field affects the CDR information. For central office (CO), direct inward and outward dialing (DIOD), foreign exchange (FX), and wide area telecommunications service (WATS) trunk groups, the **Analog PPM** field affects the CDR information.

4 In the **Disconnect Supervision-In** field, perform one of the following actions:

- Type **y** if you want:
 - Trunk-to-trunk transfers that involve this trunk group
If you want trunk-to-trunk transfer in your system, you must also set the **Transfer** field on the **Feature-Related System Parameters** screen to **y**.
 - To make the far end server or switch responsible for releasing the trunk, when the far end server sends a release signal as the calling party releases an incoming call.
 - To enhance Network Call Redirection
- Type **n** if:
 - You do not want trunk-to-trunk transfers that involve this trunk group.
 - The far end server does not provide a release signal.
 - The hardware in your system cannot recognize a release signal.
 - You prefer to use timers for disconnect supervision on incoming calls.

The system displays the **Disconnect Supervision-In** field if the **Direction** field is set to **incoming** or **two-way**.

If the **Direction** field is set to **outgoing**, the system sets the **Disconnect Supervision-In** field to **n**.

The value in the **Disconnect Supervision-In** field determines whether the system receives disconnect supervision for incoming calls over this trunk group.



CAUTION:

The system does not allow trunk-to-trunk transfers unless at least one party on the call can provide disconnect supervision. If you administer the **Disconnect Supervision-In** field incorrectly, you can cause trunks to become unusable until the problem is detected and the trunks are reset.

For example, if a user connects two trunks through the use of the Conference feature or the Transfer feature, and a far end media server on the resulting connection does not provide disconnect supervision, the trunks are not released. The trunks are not released because the system cannot detect the end of the call.

Usually, the COs in the United States provide disconnect supervision for incoming calls, but do not provide disconnect supervision for outgoing calls. Public networks in most other countries do not provide disconnect supervision for incoming calls or outgoing calls. Talk with your network services provider to determine if the public networks in your area provide disconnect supervision.

- 5 Press **Enter** to save your changes.

Administering CDR for a data module

To administer CDR for a data module:

- 1 Type **change data-module *n***, where *n* is the number of the data module for which you want to administer CDR. Press **Enter**.

The system displays the **Data Module** screen ([Figure 96, Data Module screen](#), on page 440) and ([Figure 97, Data Module screen](#), on page 441).

Figure 96: Data Module screen

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

Figure 97: Data Module screen

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	

The system displays the Ext and Name display-only fields in the **ABBREVIATED DIALING** area. The fields contain the extension number and the name of the users who have associated data extension buttons, and who share this data module.

2 In the **BCC** field, perform one of the following actions:

- Type a **1** if the speed is 56 kbps.
- Type a **2, 3, or 4** if the speed is 64 kbps.

The system compares the speed setting that you assign here with the speed setting in an associated routing pattern. The system compares the two speed settings when calls that attempt to use the data module are not completed.

The system displays the **BCC** field appears if the **ISDN-PRI** field or the **ISDN-PRI Trunks** field on the **Optional Features** screen is set to **y**.

3 The **CAPABILITIES** area contains three fields.

- In the **Configuration** field, perform one of the following actions:
 - Type **y** if you want to view and change options from originating or receiving DTEs, such as non intelligent terminals.
 - Type **n** if you do not want view and change options from intelligent devices such as computers.

The system displays the **Configuration** field only when the **KYBD Dialing** field on the **Data Module** screen is set to **y**.

- In the **KYBD Dialing** field, perform one of the following actions:
 - Type **y** if you want the users to dial calls from a keyboard, and to allow the data module endpoint to transmit and receive text during call origination or call termination.

If you type **y**, you must also type **n** in the **Low** field in the **SPEEDS** area of the **Data Module** screen.
 - Type **n** if you do not want the users to dial calls from a keyboard.

If you type **n**, the data module endpoint cannot transmit and receive text during call origination and call termination.

If you type **n**, data calls can be answered, but text feedback is not provided.

- In the **Busy Out** field, perform one of the following actions:
 - Type **y** if you want the system to place the data line circuit (DLC) port in a busyout state so calls do not terminate at the data terminal equipment (DTE) when the DTE control lead to the DLC drops. Use this option for DTEs that are members of a hunt group.
 - Type **n** if you want the system to keep the DLC port out of a busyout state when the DTE control lead to the DLC drops.

4 The **CIRCUIT SWITCHED DATA ATTRIBUTES** area contains information that is used with 7500 data modules and World Class BRI data modules.

Note that the fields in the **CIRCUIT SWITCHED DATA ATTRIBUTES** area contain default information. The default information is for modem pooling conversion resource insertion when the endpoint does not support the data query capability or the administered connections. The information in the fields has no significance for data modules that provide data query, such as Avaya-supported ISDN-BRI data modules. Use the system default settings for Avaya ISDN-BRI data modules or World Class ISDN-BRI data modules.

- In the **Default Duplex** field, perform one of the following actions:
 - Type **full** to allow simultaneous, two-way transmission.,which is duplex mode.
 - Type **half** to allow only one transmission direction at a time, which is half-duplex mode.
- In the **Default Mode** field, perform one of the following actions:
 - Type **sync** for synchronous data mode.
 - Type **async** for asynchronous data mode.
- In the **Default Speed** field, type the data rate. The valid entries are:
 - **1200**
 - **2400**
 - **4800**
 - **9600**
 - **19200**
 - **56000** when the **Default Mode** field is set to **sync**
 - **64000** when the **Default Mode** field is set to **sync**

5 The system displays the **Connected To** field, when the **Type** field contains either **dpm** or **data-line**.

In the **Connected To** field, perform one of the following actions:

- Type **dte** if the Asynchronous Data Unit (ADU) is connected to a data terminal equipment (DTE).
- Type **isn** if the ADU is connected to an information systems network.

6 In the **COR** field, type the number of the Class of Restriction (COR) for this data module. Valid entries are **0** through **95**.

7 In the **COS** field, type the number of the Class of Service (COS) for this data module. Valid entries are **1** through **15**.

- 8 The DATA MODULE CAPABILITIES area contains three fields with information for the 7500 data modules and the World Class BRI (WCBRI) data modules.

- The **Default Data Applications** field identifies the mode that the system uses to originate calls when the calling parameters do not specify the mode. The system also uses the mode to terminate trunk calls that do not have administered connections, or for which the bearer capability is unspecified. For more information, see the “Uniform Dial Plan” feature.

In the **Default Data Applications** field, perform one of the following actions:

- Type **M0** to specify mode 0. Use this option for a WCBRI endpoint that the system uses as an administered connection.
 - Type **M1** to specify mode 1.
 - Type **M2_A** to specify mode 2 asynchronous.
 - Type **M2_S** to specify mode 2 synchronous.
 - Type **M3/2** to specify mode 3/2 adaptable.
- In the **Default ITC** field, perform one of the following actions:
 - Type **restricted** for a WCBRI endpoint that is an administered connection.
 - Type **unrestricted** for a WCBRI endpoint that is not an administered connection.
 - The display-only **MM Complex Voice Ext** field contains the number of the associated telephone in the multimedia complex. The system displays the **MM Complex Voice Ext** field only when the **Multimedia** field is set to **y**.

The field is blank until you type the data module extension in the **MM Complex Data Ext** field on the **Station** screen. When you type the data module extension in the **MM Complex Data Ext** field on the **Station** screen, the system associates the numbers in the **MM Complex Data Ext** and the **MM Complex Voice Ext** fields as two parts of a one-number complex. The one-number complex is the extension of the telephone.

The system displays the data module extension in the display-only **Data Extension** field.

- 9 In the **ITC** field, perform one of the following actions:

- Type **restricted** if the data module can send bits at speeds less than or equal to 56 kbps. If you type **restricted** in the **ITC** field, the system uses a trunk group for which the **COMM Type** field on the **Trunk Group** screen is set to **rbavd** or **avd** to complete a call from this data module endpoint.

A restricted transmission facility enforces ones density digital transmission. Ones density digital transmission is a sequence of eight digital zeroes that the firmware on the DS1 port board converts to a sequence of seven zeroes and a digital 1.

- Type **unrestricted** if the data module can send bits at a speed up to, and including, 64 kbps. If you type **unrestricted** in the **ITC** field, the system uses a trunk group for which the **COMM Type** field on the **Trunk Group** screen is set to **avd** to complete a call from this data module endpoint. The value **avd** in the **Comm Type** field indicates that the trunk group provides both restricted and unrestricted transmission facilities.

An unrestricted transmission facility does not enforce ones density digital transmission. The DS1 port board firmware does not convert the digital information.

The system does not display the **ITC** field for voice-only stations or BRI stations. The **ITC** field applies only when the **Comm Type** field on the **Trunk Group** screen, that the system uses for an outbound call, is set to **avd** or **rbavd**. The **ITC** field specifies the type of transmission facilities that an ISDN call uses when a call originates from this data module endpoint.

10 In the **List1** field in the **ABBREVIATED DIALING** area, perform one of the following actions:

- Leave the field blank if you do not want the data module to have an abbreviated dialing list.
- Type **e** if you want the data module to have an enhanced abbreviated dialing list.
- Type **g** if you want the data module to have a group list.

If you type **g**, the system displays a field to the right of the **List1** field. You must type a group list number in this field.

- Type **p** if you want the data module to have a personal list.

If you type a **p**, the system displays a field to the right of the **List1** field. You must type a personal list number in this field.

- Type **s** if you want the data module to have a system abbreviated dialing list.

11 In the **Name** field, perform one of the following actions:

- Type the name of the user who is associated with the data module.
- Leave the field blank.

12 The **OPTIONS** area contains six fields.

- In the **Answer Text** field, perform one of the following actions:

— Type **y** if you want the system to allow text feedback to the DTE when a user answers a call or the system disconnects a call. The text feedback includes both DLC-generated text and system-generated text.

— Type **n** if you want the system to disable text feedback to the DTE when a user answers a call or the system disconnects a call, and when the DTE that answers a call is a computer or an intelligent device. The system still generates the text, but the DLC does not allow the text to be delivered to the DTE.

The **Answer Text** field applies to the following call messages:

- Incoming
- Answered
- Disconnected
- Disconnected other end

The system displays the **Answer Text** field only if the **KBDY Dialing** field on the **Data Module** screen is set to **y**.

- In the **Connected Indication** field, perform one of the following actions:

— Type **y** if you want the system to generate a “connected” message to the DTE when the system establishes a connection.

— Type **n** if you do not want the system to generate a “connected” message to the DTE when the system establishes a connection.

The system displays the **Connected Indication** field only if the **KBDY Dialing** field on the **Data Module** screen is set to **y**. If the **Connected Indication** field is set to **n**, DLC provides the connection indication when the DLC activates the EIA 232C control lead.

- In the **Dial Echoing** field, perform one of the following actions:

— Type **y** if you want the system to echo characters back to the DTE.

— Type **n** if you do not want the system to echo characters back to the DTE and when an intelligent device provides keyboard dialing.

The system displays the **Dial Echoing** field only if the **KBDY Dialing** field on the **Data Module** screen is set to **y**.

- In the **Disconnect Sequence** field, perform one of the following actions:
 - Type **long-break** if you want a break that is greater than 2 seconds.
 - Type **two-breaks** if you want a break that is less than 1 second.

The system displays the **Disconnect Sequence** field only if the **KBDY Dialing** field on the **Data Module** screen is set to **y**.

- In the **Parity** field, type one of the following types of parity:
 - **even**
 - **odd**
 - **mark**
 - **space**

The system displays the **Parity** field only if the **KBDY Dialing** field on the **Data Module** screen is set to **y**. The DLC generates the parities when the DLC sends call setup text to the DTE. The DLC does not check the parity when the DLC receives dial characters. Select the parity that matches the DTE that connects to the data module.

- In the **Permit Mismatch** field, perform one of the following actions:
 - Type **y** if you want the DLC to operate at the highest selected speed, which is a higher rate than that of the far end data module.
 - Type **n** if you do not want DLC to operate at the highest selected speed.

The **Permit Mismatch** field contains information that allows an EIA interface to operate at a rate that differs from the rate that is agreed upon during the data module handshake. The rate that is agreed upon during the data module handshake is always the highest compatible rate among the speeds that each data module reports.

The information in the **Permit Mismatch** field eliminates the need to change the DTE or DLC speed whenever someone, or something, places a call to or from endpoints that operate at a different speed.

When the **Permit Mismatch** field is set to **y**, the DLC reports the highest optional speed and all the lower speeds, or the previously selected autoadjust speed, during the handshake process.

- 13** In the **Port** field, type the appropriate values from [Table 47, Port field values](#), on page 445.

Table 47: Port field values

Characters	Description	Value
1-2	cabinet number	01 through 44 (For DEFINITY R configurations) 01 through 03 (For DEFINITY SI configurations) 01 through 64 (For S8700 IP-Connect)
3	carrier	A through E
4-5	slot number	0 through 20
1 of 2		

Table 47: Port field values

Characters	Description	Value
6-7	circuit number	01 through 04 (x.25 circuit pack) 01 through 31 (DEFINITY SI, S8700 IP-Connect (tdm, pdm) configurations) 01 through 16 (ppp for S8700 IP-Connect) 01 through 08 (system-port for S8700 IP-Connect) 17/33 (Ethernet on S8700 IP-Connect)
x	Administration without Hardware	If the Secondary data module? field, contains an n , you can type x in the Port field. A Port field set to x indicates that no hardware is associated with the port assignment.
2 of 2		

14 The SPEEDS area contains information about the operating speeds of the data module.

- In the Low field, perform one of the following actions:
 - Type **y** if you want the data line circuit to operate at a speed of 0 to 1800 bps.
 - Type **n** if the KYBD Dialing field on the **Data Module** screen is set to **y**.
- In the 300, 1200, 2400, 4800, 9600, and 19200 fields, perform one of the following actions:
 - Type **y** if you want the DLC to operate at the speed.
 You can choose any of the speeds for the DLC. The DLC matches the speed for the duration of the call.
 If you select multiple speeds, you must also set the Autoadjust field to **n** and select at least three speeds. The speed of the DTE must be the highest speed that you select. The DTE must have the highest speed because the system delivers feedback to the DTE at the highest selected speed.
 - Type **n** if you do not want the DLC to operate at the speed.

In the Autoadjust field, perform one of the following actions:

- Type **y** if you want the DLC port to automatically adjust to the operating speed and the parity of the DTE to which the DLC port connects.
- Type **n** if you do not want the DLC port to automatically adjust to the operating speed and the parity of the DTE to which the DLC port connects.

The system displays the Autoadjust field when the KYBD Dialing field on the **Data Module** screen is set to **y**. The Autoadjust field applies only to calls that a user originates from a keyboard.

15 In the SPECIAL DIALING OPTION field, perform one of the following actions:

- Leave the field blank if you do not want the data module to have special dialing
- Type **hot-line**, if you want the data module to have hot-line dialing.

If you type **hot-line**, the system displays the Abbreviated Dialing Dial Code (from above list): field. Type the abbreviated dial code in the Abbreviated Dialing Dial Code (from above list): field. Valid entries are **0** through **999**.

- Type **default**, if you want the data module to have default dialing.

If you type **default**, the system displays the **Abbreviated Dialing Dial Code (from above list):** field. Type the abbreviated dial code in the **Abbreviated Dialing Dial Code (from above list):** field. Valid entries are **0** through **999**.

16 In the **TN** field, type the tenant partition number of the data module. Valid entries are **1** through **100**.

17 In the **Type** field, perform one of the following actions:

- Type **7500** to assign a 7500 data module.

The 7500 data module supports:

- Automatic TEI
- B-channel, maintenance and management messaging
- Service Profile Identifier (SPID) initialization capabilities.

BRI voice endpoints, BRI data endpoints, or both BRI voice and BRI data endpoints 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 circuit pack supports up to 12 ports.

BRI provides a multipoint capability. Therefore, you can administer more than one ISDN endpoint, either a voice endpoint or a data endpoint, on one port.

For BRI, multipoint administration allows for telephones that have SPID initialization capabilities. Multipoint administration is allowed only if no endpoint that is administered on the same port is a fixed tie endpoint, and no station on the same port has B-channel data capability. The system restricts multipoint administration to two endpoints per port.

- Type **data-line** to assign a data line data module.

Use the **Data Line Data Module (DLDM)** screen to assign ports on the Data Line (DLC) circuit pack that allow EIA 232C devices to connect to the system. The DLC, with a companion 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 of 300, 1200, 2400, 4800, 9600, and 19200 bps over two-pair, full-duplex lines. These lines can have different lengths, depending on the transmission speed and the wire gauge.

The DLC has eight ports. The connection from the port to the EIA device is direct, which means 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 DTE, and as a digital line port to the server that runs Communication Manager.

The DLC connects the following EIA 232C equipment to the system:

- Printers
- Non intelligent data terminals
- Intelligent terminals and personal computers (PCs)
- Host computers
- Information Systems Network (ISN), RS-232C local area networks (LANs), or other data switches

- Type **pdm** to assigns a DCE interface for processor/trunk data modules.

Use the **Processor/Trunk Data Module** screen to assign Modular Processor Data Modules (MPDMs) and Modular Trunk Data Modules (MTDMs). Use one screen to assign MPDMs (700D), 7400B, 7400D or 8400B Data Module. Use another screen for MTDMs (700B, 700C, 700E, 7400A). You must complete one screen for each MPDM, 7400B, 7400D, 8400B or MTDM.

The MPDM, 7400B, or 8400B Data Module provides a Data Communications Equipment (DCE) interface. Use the interface for a connection to equipment such as a data terminal, call detail recording (CDR) output device, on-premises administration terminal, Message Server, Property Management System (PMS), AUDIX, and host computers. The MPDM, 7400B, or 8400B Data Module also provides a Digital Communications Protocol (DCP) interface to the digital switch. Note that DCE is the equipment on the network side of a communications link that provides all the functions that are required to make the binary serial data from the source or transmitter compatible with the communications channel.

The MTDM provides an EIA 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. Note that DTE is the equipment that comprises 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 the 7400A Data Module can also serve as part of a conversion resource for combined modem pooling.

- 18** Press **Enter** to save your changes.

Identifying the Inter Exchange Carrier

To identify the Inter Exchange Carrier (IXC) in the CDR record:

- 1** Type **change ixc-codes**. Press **Enter**.

The system displays the **Inter-Exchange Carrier Codes** screens ([Figure 98, Inter-Exchange Carrier Codes screen](#), on page 448) and ([Figure 99, Inter-Exchange Carrier Codes screen](#), on page 449).

Figure 98: Inter-Exchange Carrier Codes screen

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:	_____	_____			

Figure 99: Inter-Exchange Carrier Codes screen

change ixc-codes		Page 2 of 2	
	IXC Prefix	IXC Code Format	
1.	_____	_____	
2.	_____	_____	
3.	_____	_____	
4.	_____	_____	
5.	_____	_____	

- 2** In the **IXC Access Number** field, type the digits that the user dials, or that AAR/ARS inserts, into the outputted digit string so the system can access the IXC. The system does not allow duplicate access numbers in the **Inter-Exchange Carrier Codes** screen.

The system accepts a string of 2 to 11 numbers or the asterisk character (*) in the **IXC Access Number** field.

- 3** In the **IXC Name** field, type a description of the IXC. The system accepts a 1 to 15 character string in the **IXC Name** field. You can also leave this field blank.

- 4** In the **IXC Code Format** field, type the format for the IXC code. The valid entries for this field are a:

- 1 to 4 digit code
- Asterisk (*)
- Lower-case x
- Upper-case X
- String of 4 lower-case x's (xxxx) to indicate line one
- String of 3 lower-case x's (xxx) to indicate line two

- 5** In the **IXC Prefix** field, type the prefix for the IXC code. The valid entries for this field are a:

- Prefix of 1 to 3 digits
- Asterisk (*)
- 101 to indicate line one
- 10 to indicate line two

- 6** In the **CDR Account Code Access Code** field, type the access code that a user enters before the user enters a CDR account code.

- 7** Press **Enter** to save your changes.

Administering CDR for the paging ports

To administer CDR for the paging ports:

- 1 Type **change loudspeaker paging**. Press **Enter**.

The system displays the **Loudspeaker Paging** screen ([Figure 100, Loudspeaker Paging screen](#), on page 450).

Figure 100: Loudspeaker Paging screen

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:	_____	_____	_____	_____	_____	_____	_____	_____			

- 2 In the CDR field, perform one of the following actions:
 - Type **y** if you want the system to record CDR information on the paging ports.
 - Type **n** if you do not want the system to record CDR information on the paging ports.
- 3 Press **Enter** to save your changes.

Administering the Intra-Switch CDR

To administer Intra-Switch CDR:

- 1 Type **change intra-switch-cdr**. Press **Enter**.

The system displays the Intra-Switch CDR screen ([Figure 101, Intra-Switch CDR screen](#), on page 451).

Figure 101: Intra-Switch CDR screen

change intra-switch-cdr				Page 1 of 2	
INTRA-SWITCH CDR					
Assigned Members:		8	of 5000	administered	
1: 70103	19:	37:	55:	73:	91:
2: 70105	20:	38:	56:	74:	92:
3: 70112	21:	39:	57:	75:	93:
4: 70115	22:	40:	58:	76:	94:
5: 70120	23:	41:	59:	77:	95:
6: 70122	24:	42:	60:	78:	96:
7: 70124	25:	43:	61:	79:	97:
8: 70126	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:

- 2 In the Assigned Members field, type the local extensions that you want to track with Intra-Switch CDR. The number of extensions that you can track can vary.
- 3 Press **Enter** to save your changes.

End-user procedures for Call Detail Recording

End-users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Call Detail Recording:

- Associating an account code with a call

To associate an account code with a call:

- 1 Dial the CDR account code access code that you assigned.
- 2 Dial an account code.
- 3 Dial a trunk access code or an access code.
- 4 Dial the telephone number.

Reports for Call Detail Recording

The following reports provide information about the Call Detail Recording feature:

- None

Considerations for Call Detail Recording

This section provides information about how the Call Detail Recording (CDR) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Call Detail Recording under all conditions. The following considerations apply to Call Detail Recording:

- **Date and Time**

If you do not administer time of day in your system, the software does not generate CDR records.

If a call is in progress while you change the time of day information, the system does not record the duration of the call in the CDR record. Instead, the system records the numeric sequence 9999 in the CDR record. This sequence indicates that the call was in progress when you changed the time of day information.

- **Dial plan**

If the dial plan in your system supports 6-digit or 7-digit extensions, only the formats that already support calling numbers that are longer than 7 digits support 6-digit or 7-digit extensions. Such numbers include expanded, unformatted, customized, and international ISDN calling numbers.

All other calling number formats send only 5 digits. If the calling number is a 6-digit or a 7-digit extension, the system sends only the last 5 digits.

The following information applies to the port that the secondary CDR output device uses:

- The data that go to the secondary port must be the same as the data that go to the primary port. You can use the following record types for the secondary output:
 - LSU
 - Int-Direct
 - Int-Process
 - Unformatted
- If the system cannot send records to the primary CDR output device, the system discontinues sending records to the secondary port for two minutes. The secondary port must operate at the highest possible speed to prevent the loss of information.
- If the output buffer is full, the system busies out the secondary port for 2 minutes. This action 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 fewer than 400 records, or 1800 records for Release 5r and later, remain in the buffer.

Interactions for Call Detail Recording

This section provides information about how the Call Detail Recording (CDR) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Call Detail Recording in any feature configuration.

- **Abbreviated Dialing**

When a user uses either Abbreviated Dialing or a Facility Busy Indicator button to place a call, all the outpulsed digits appear on the record.

- Answer Detection

CDR starts recording call duration at the time that the circuit pack detects the answer.

Answer Detection provides more accurate call records where tone detection is possible, and Network Answer Supervision is not received.

- Attendant Console

If an attendant-assisted call uses an outgoing trunk, the system records the primary extension of the user who requests the attendant service as the calling number in the CDR record. The system records the primary extension of the user even if the attendant dialed an outside number.

Condition code 1 indicates an attendant assisted the call.

If the attendant allows through dialing, the system records the primary extension of the user who dialed the number as the calling party. Condition code 1 indicates that an attendant extended a trunk access code (TAC). Condition code 7 indicates that an attendant extended a feature access code (FAC).

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 the attendant transfers.

With attendant-assisted calls that require an account code, enter the account code before the TAC.

If the attendant redirects an incoming call to an extension, the attendant can dial an account code before the attendant dials the extension number.

There are no intraswitch-optional attendant calls. However, the system generates intraswitch records for an intraswitch-optional extension call to an the attendant or for a call from the attendant to an intraswitch-optional extension. In the case of an attendant-assisted call that involves an intraswitch extension, the system records the extension of the user who called the attendant as the dialing number. The system records the extension to which the attendant extended the call as the dialed number. In this case, the record has condition code 0.

- Avaya Intuity AUDIX

The following example describes CDR for remote Intuity AUDIX over a distributed communications system (DCS). If station A on node 1 forwards calls to Intuity 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 Intuity 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 system records:

- The actual calling number in the **Calling Number** field
- The TAC of the trunk that brings the call into the local switch in the **Incoming Trunk Access Code** field of 24-word records

If the system uses an outgoing trunk when a user forwards, transfers, or conferences an incoming call, the system generates two separate CDR records. The system generates a record for incoming usage, and a record for outgoing trunk usage. The system records AUDIX as the calling number in the outgoing trunk usage record

If Incoming Trunk Call Splitting is enabled, and Transfer out of AUDIX is used, CDR generates two records. The first record contains AUDIX, the second record contains the transferred-to party.

- Authorization Codes

The system records authorization codes in CDR records as follows:

The system does not record authorization codes in the 59-character CDR International Processing and International Direct records.

- Automatic Selection of Direct Inward Dialing (DID) Numbers

If the system records an incoming call, the system records the DID extension number, not the room extension number.

- Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS)

For ARS calls, the system records that the:

- ARS call was made
- Calling extension number
- Facilities restriction level (FRL) of the calling extension
- Called number
- TAC of trunk group that is used for the ARS call
- Time of call completion
- Call duration
- Inter-exchange carrier (IXC) code, if any

The system does not generate a CDR record if CDR is suppressed for the trunk group that ARS uses. If CDR is not suppressed, the system generates condition code 7. The system records the following information:

- The ARS access code in the **Access Coded Dialed** field
- The TAC for the trunk group that the call used 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 the reorder tone and the CDR record shows an ineffective call attempt.

If an ARS call is an attendant-assisted call, the CDR record shows the call with condition code 7 instead of condition code of 1. Condition code 7 indicates an ARS call. Condition code 1 indicates an attendant-assisted call. The system generates these condition codes, 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.

With the Forced Entry of Account Codes (FEAC) capability, the system does not use the class of service (COR) of the trunk group to determine if the user must enter an account code, if the system uses ARS to access the trunk.

- Automatic Callback

When the Automatic Callback feature is used for an intraswitch call, the system does not generate a CDR record for the first call attempt, nor for the ringback. However, if intraswitch is enabled for either the calling user or the called user, the system generates a CDR record of the call, if the called user answers and completes the call.

- Automatic Circuit Assurance (ACA)

ACA calls generate intraswitch CDR records if CDR monitors the terminating extension. The originating extension for ACA calls cannot be administered for intraswitch monitoring.

- Automatic Wakeup

The system does not generate CDR intraswitch records for wake-up calls.

- Bridged Call Appearance

The system does not record CDR information about the user who bridges onto a call. Instead, the system records the number that a user dialed in the **Dialed Number** field of the CDR record. The system records the duration of the call when the last party drops off the call.

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**

Attendants and users do not need to enter an account code to make a busy verification.

- **Call-by-Call Service Selection**

When the system successfully makes a call on a Call-by-Call Service Selection trunk, the system translates the network-specific facility that the system uses for the call into an ISDN network service (INS) number. The system records the INS number in the INS field of the CDR record.

If the system is unsuccessful in making a call on a Call-by-Call Service Selection trunk because of an administered trunk usage allocation plan, the system records the INS number in the INS field of the CDR Record with condition code *E*.

- **CallVisor Adjunct-Switch Application Interface (ASAI)**

Call classification competes with CallVisor ASAI switch-classified calls for ports on the call classifier circuit pack. Answer Detection sends a report of a connect event to ASAI.

- **Call Coverage**

If the system does not route a call to an off-network coverage point, the system records the extension number that the calling user dials as the dialed number when a user answers an incoming call or an intraswitch call at a covering extension.

If the system routes a call to an off-network coverage point, the system records the number at the off-network location as the dialed number. The system records the extension that has the off-network location in its coverage path as the calling number.

- **Call Forwarding All Calls**

If the system does not forward a call to an off-network location, the system records the number that the user dials as the dialed number.

If the system forwards a call to an off-network location, the system records the number of the off-network location as the dialed number. The system records the extension from which the call was forwarded as the calling number.

The system generates one CDR record for a forwarded intraswitch call. The dialed number in the record is the extension that the calling user dialed.

The system generates two CDR records for a trunk call to a station that the system forwards to another trunk. The first record shows an incoming trunk call to the station. The second record shows an outgoing trunk call from the station.

With the FEAC capability, the system cannot forward to a destination at which a user must enter an account code.

- **Call Park**

When a user parks an incoming call or an intraswitch call, the system records the extension of the user as the dialed number in the CDR record. The system records the entire time that the incoming trunk is busy as the duration of the incoming call. The system records the time that the call started until the call ends as the call duration for an intraswitch call.

- **Call Pickup**

The system records the number that the user dials as the dialed number when a member of a pickup group answers an incoming call or an intraswitch call.

- **Call Prompting**

Call classification competes with Call Prompting for ports on the call classifier circuit pack.

- Call Vectoring

You can administer CDR so the system records the vector directory number (VDN) extension instead of the extension of the Hunt Group or of the member. If you administer CDR so the system records the VDN extension, you override the Calls to Hunt Group - Record option of CDR for incoming call vectoring calls.

Outgoing vector calls generate ordinary outgoing CDR records. The system records the originating extension as the calling number.

The system records the duration of the call from the time that answer supervision is returned for incoming calls to a VDN.

- If the vector returns answer supervision, and the call does not go to another extension, the system records the VDN extension as the called number in the CDR record. The vector can return answer supervision with an announcement, collect, disconnect, or wait with music command.
- If the call terminates to a hunt group, the system records the VDN, the hunt group extension, or the agent extension as the called number in the CDR record.
- If the call terminates to a trunk, CDR generates an:
 - Incoming record with the incoming TAC as the dialed number.
 - Outgoing record with the incoming TAC as the calling number and the digits that are dialed through the vector step as the dialed number.

If you administer member extension for CDR, the system records an incoming call to the station if the system successfully routes a call to the station with the **route-to** command.

The system does not generate ineffective call attempt records for unsuccessful Call Vectoring **route-to** commands.

If a vector interacts with an extension or a group that has Call Forwarding All Calls active, normal Call Forwarding and CDR interactions apply.

Some calls look like intraswitch calls. Such calls include, for example, a call for a station that is administered for intraswitch CDR to a VDN, which becomes an outgoing call on an outgoing trunk. The system does not generate intraswitch CDR records for calls that look like intraswitch calls, but that are not intraswitch calls. The system generates a record with condition code A, which indicates outgoing.

- Call Waiting Termination

The system starts the call duration timer when a user answers an incoming call.

- Centralized Attendant Services (CAS)

The system records the extension of the user who originates a call as the originator of a call, if a CAS attendant extends the call to the user, and CDR is not assigned to the release-link trunk (RLT) trunk group.

The system records the RLT trunk as the originator of a call, if a CAS attendant extends the call to the user and CDR is assigned to the release-link trunk (RLT) trunk group.

The system does not generate a CDR record, if a CAS attendant answers a call but does not extend the call to a user.

- CO Trunks

The system records all incoming and outgoing calls on a central office (CO) trunk group, if CDR is assigned to the trunk group, and CDR is administered to record incoming calls.

- Conference

The system records a conference call for CDR, if either of the following conditions is met:

- The call uses at least one trunk that is eligible for CDR, and has two or more parties
- The call has at least one party that is optioned for intraswitch CDR.

The system records condition code *C* for each conference call CDR record.

The system generates a separate conference call CDR record for each outgoing trunk and each incoming trunk that serves the conference call. If you enable either, the system also generates a separate record for each internal party on the call.

For the outgoing portion of a conference call that involve multiple extensions, the system records the extension of the user who requests the outside dial tone to include another participant, as the calling party.

The system records the entire time that an incoming trunk or an outgoing trunk is used for a conference call, as the duration of the call.

The system generates a separate CDR record for each trunk that is used in a trunk-to-trunk transfer. If incoming trunk call splitting (ITCS) is active, the incoming trunk record shows the duration of the entire call.

The system starts a new CDR record whenever the originator of a conference call dials a nontrunk participant, if the conference call is optioned for intraswitch CDR. For example, station 1 is optioned for intraswitch CDR and calls station 2. Station 1 includes station 3 in the conference call. Station 1 drops from the call. Station 2 or station 3 drops from the call. The system generates two CDR records with condition code *C*. The system generates one record from station 1 to station 2, and another record from station 1 to station 3.

The system generates one record with condition code *C* for each dialed intraswitch conference participant, if any of the conference participants are optioned for intraswitch CDR. The system generates a record with condition code *C*, even if the originator of the conference is not optioned for intraswitch CDR. For example, station calls station 2, which is optioned for intraswitch CDR. Station 1 includes station 3 into the conference call. Station 1 drops from the call. Station 2 or station 3 drops from the call. The system generates one CDR record with condition code *C* from station 1 to station 2.

The system generates intraswitch conference call CDR records when both the calling party and the called party call drop from the call. The system records the call duration from the time that the called party answers the call until both the calling party and the called party drop from the call.

If an attendant originates a conference, the system generates CDR records only for the dialed numbers that correspond to any intraswitch optioned extensions.

- Distributed Communications System DCS

The system does not pass station information throughout the DCS network for CDR records.

- Direct Inward Dial (DID) trunks

If you administer the system to record incoming CDR information, and if you administer CDR for the trunk group, the system records all incoming calls on the DID trunk group.

- Emergency Access to the Attendant

The system does not generate intraswitch CDR records for Emergency Access calls.

- Expert Agent Selection (EAS)

You can assign a logical extension to an agent who can then use the logical extension to log in to a telephone. You can administer CDR so the system records the logical extension of the agent as the called number. The system records the logical extension as the called number instead of the extension of the hunt-group or the hunt group member.

- Foreign Exchange (FX) Trunks

If you want the system to generate CDR records for calls to FX trunks, you must administer your system to generate those records. You must also administer each trunk group so the system generates a CDR record for the trunk group.

- Hotline Service

The system generates a CDR record of the stored number that is used on an outgoing or intraswitch Hotline call as if someone manually dialed the number.

- Hunt Groups

You can administer CDR so the system records either the extension of the hunt group or the extension of the individual hunt group member as the called number.

- Intercept Treatment

If the system routes an outgoing call or a tandem call to intercept treatment, the system records the number that the user dialed as the dialed number. The system also records condition code *F*.

- Inter-PBX Attendant Calls

[Table 48, Inter-PBS CDR information](#), on page 458 shows the information that the system records if a user calls an Inter-PBX attendant, and the call uses a trunk group that has CDR assigned.

Table 48: Inter-PBS CDR information

CDR data field	Value
Condition Code	A
Access Code Dialed	Blank
Access Code Used	The TAC of the trunk that the call used
Dialed Digits	The Inter-PBX attendant access code

- ISDN

The system sends an indication to the CDR device when the system receives a true answer supervision.

The system creates a CDR record each time that the system networks an ISDN call. In this case, the answer supervision information that the system records might not be accurate. If you use unformatted or expanded record formats, the station identification number (SID) or automatic number identification (ANI) appears in the CDR record, if the SID or ANI is sent.

- Last Number Dialed

The system stores the CDR access code and the account code that the user dials as part of the Last Number Dialed. However, some digits might be lost, because of the limit on the number of digits that can be stored for the Last Number Dialed feature.

- **Manual Originating Line Service**

If an attendant establishes an outgoing call for a user, and designates the call as a Manual Originating Line call, the system generates an attendant-assisted outgoing call CDR record. The system records the extension of the user who originated the call as the calling number, and applies condition code 1.
- **Multiple Listed Directory Numbers (LDNs)**

If the system records incoming call information, the system records the extension number or the TAC to which the attendant completes the call, as the called number of LDN calls.

If the system records incoming call information, the system records the attendant extension as the dialed number, if the call terminates at the attendant console.

You cannot administer LDNs for intraswitch CDR. However, the system generates an intraswitch CDR record for a call from an intraswitch-optional extension to an LDN.
- **Night Service**

The system records the extension number that is assigned to the attendants as the dialed number for night service calls.
- **Off-Premises Station**

The system generates a CDR record for a call to an off-premises station when the:

 - Call involves an outgoing or an incoming trunk call.
 - Off-premises station is optional for intraswitch CDR.
 - Other terminal that is involved in the call is optional for intraswitch CDR.
- **Personal Central Office Line (PCOL) trunks**

If the system records incoming calls, the system records the primary extension of the user who answers the call as the called number for incoming PCOL call.

The system records an outgoing PCOL call as a call from the originating extension number through the trunk group that is associated with the PCOL.

The system records the dialed number in the **Dialed Number** field for an outgoing PCOL call. The system does not record the TAC in the **Dialed Number** field for an outgoing PCOL call.
- **Planned Interchange**

When any planned interchange occurs, the system might record calls that end within 10 to 20 seconds after the interchange as calls that have an invalid duration. A call with an invalid duration is a call with a duration of 9:59:9, and a condition code other than 4. These call records are invalid. Deviations in the clocks between the two processors, and the short duration of the calls, cause the invalid duration.
- **Private Network Access**

The system records Private Network Access calls, if CDR is assigned for incoming or outgoing tie trunks.
- **Remote Access**

The system records remote access calls if Remote Access is provided on a per-trunk-group basis, and you administered CDR for those trunks. The trunk group access code in the call record is the only indication that the record is for a remote access call.

- **Ringback Queuing**

The system records condition code 8 for an outgoing call that is in a trunk queue before the call is complete. The system does not record the time that the call is in the queue.

The system does not generate a CDR record if a call waits in a trunk queue, and the call is not successfully completed. The call is not completed successfully if the time that the call waits in the queue exceeds the wait limit for the queue, or if the calling party does not answer the callback.
- **Security Violation Notification (SVN)**

The system generates a CDR record for a SVN call if the terminating extension is monitored. You cannot administer the originating extension for intraswitch monitoring.
- **Service Observing**

The system does not generate CDR records for Service Observing calls.
- **Tandem Tie-Trunk Switching**

The calling party on an incoming trunk can dial the CDR account code. The system records the TAC for the incoming trunk group in the **Calling Number** field in the CDR record. The system records the number that the user dials as the number dialed.
- **Temporary Bridged Appearance**

A CDR record is not affected if a second user, or subsequent user, bridges a call.
- **Temporary Signaling Connections (TSCs)**

If you administer CDR to use ISDN layouts, the system records call-associated TSCs and TSC requests in the call record. If you administer CDR to record noncall-associated/temporary-signaling connection (NCA TSCs) and TSC requests, the system generates separate CDR records for each type of TCS. The system records the TCS data in the **TSC Flag** field and **Packet Count** field.
- **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 system records the originating extension as the calling party.

If a user receives a call on an incoming trunk and then transfers the call to another extension, the system records the extension that originally received the call as the dialed number.

If a user receives an intraswitch call and then transfers the call to another extension, the system records the extension that originally received the call as the dialed number.

The system generates two CDR records if all the following conditions are met:

 - Call splitting is active.
 - A user receives or originates a trunk call.
 - The user transfers the call to another extension

The system generates intraswitch CDR records 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 station C drops, the system generates two CDR records with a condition code 0. The system generates a CDR record for a call from station A to station B, and a second record for the call from station A to station C.

The system generates intraswitch CDR transfer records when both the calling party and the called party drop from the call. The system records the call duration from the time that the called party answers the call until both the calling party and the called party drop the call.

The system generates an incoming trunk call CDR record if ITCS is enabled, and a user transfers the call to a local extension that is optioned for Intraswitch CDR. The system does not generate an intraswitch record.

When a user transfers a call to another extension, a users cannot dial an account code, unless the user has console permissions.

When a user transfers a call to a trunk, the user can dial an account code before the user dials the ARS or the TAC.

- Trunk-to-Trunk Transfer

With CDR, the system processes a Trunk-to-Trunk Transfer connection as a conference call. The system generates a separate CDR record for each trunk in the connection.

You can administer CDR so the system records unanswered trunk calls. You can administer each trunk group so the system records unanswered calls if the calls remain unanswered for an interval that you specify.

If Incoming Trunk Call Splitting is active, the system generates a CDR record for a trunk-to-trunk transfer. The system generates a record of the incoming call, and a record of the outgoing call. The system records the duration of the outgoing call from the time that the user transfers the call until both parties drop the call. The system records the duration of the incoming call from the time that the user answers the call until both parties drop the call.

- Uniform Dial Plan (UDP)

[Table 49, Uniform dial plan CDR information](#), on page 461 shows the information that the system records if a user uses a UDP extension to call another user, and the trunk group that the call uses has CDR assigned.

Table 49: Uniform dial plan CDR information

CDR data field	Value
Condition Code	7
Access Code Dialed	Blank
Access Code Used	The TAC of the trunk that the call used
Dialed Digits	The UDP extension

- VDN Return Destination

The system does not generate a CDR record for an incoming call until the originator drops from the call. The system creates a CDR record when all the following conditions are met:

- A call goes to the return-destination VDN.
- The originator has not dropped.
- Vector processing, that is the return destination VDN, routes the call to an outgoing trunk.

The system does not create a CDR record if vector processing routes a call from the return-destination VDN to an internal call. The system records only the first VDN that the caller accesses, regardless of the number of other extensions that are involved in the call.

If the system routes an incoming VDN call to a station, the system includes the station in the CDR record. If the system routes an incoming VDN call to an outgoing trunk, the system includes the VDN in the CDR record.

Call Forwarding

Use the Call Forwarding feature to redirect calls to:

- An internal extension
- An off-network number
- An attendant group

Call Forwarding supports the following capabilities:

- **Call Forwarding All Calls**
Users use the Call Forwarding All Calls capability to redirect every incoming call to another destination.
- **Call Forward Busy/Don't Answer**
Users use the Call Forward Busy/Don't Answer capability to redirect incoming calls to another destination when the user is busy on a call, or does not answer a call within the allowed time interval.
- **Call Forwarding Off Net**
Users use the Call Forwarding Off Net capability to forward calls to an off-network destination. If the Coverage of Calls Redirected Off-Net (CCRON) capability is active, the system monitors a call for call progress tones. If no one answers the call at the off-network destination, the system returns the call to the internal extension.
- **Call Forwarding Override**
Users use the Call Forwarding Override capability to call a user, or transfer a call to a user, who has the Call Forwarding Feature active. The user can use the Call Forwarding Override capability only if the user extension is the destination of the forwarded calls.

Detailed description of Call Forwarding

This section provides a detailed description of the Call Forwarding feature.

Call Forwarding All Calls

Users use the Call Forwarding All Calls capability to redirect any incoming calls to another destination. You can restrict access to the Call Forwarding All Calls capability to specific users.

A user cannot have both the Call Forwarding All Calls capability and the Call Forward Busy/Don't Answer capability active at the same time.

The system forwards a call only once. For example, assume that extension A designates extension B as its forwarded-to destination, and that extension B designates extension C as its forwarded-to destination. When someone calls extension A, the system:

- Rings the call at extension A, if possible
- Rings the call at extension B, if possible

Call Forwarding

Detailed description of Call Forwarding

- Redirects to the coverage path of extension A, if a coverage path is available at extension A, and if the coverage criteria of extension A are satisfied when applied at extension B
- Does not forward to extension C under any circumstances

The system can forward an unlimited number of calls simultaneously.

Call Forwarding and FAC

Users use a Feature Access Code (FAC) or a Call Forward-All feature button to activate or deactivate Call Forwarding All Calls for their own telephone. Virtual extension users cannot activate or deactivate Call Forwarding All Calls. Users can activate or deactivate the Call Forwarding All Calls feature for the following entities:

- Another extension
- A virtual extension
- An Automatic Call Distribution (ACD) split

Call Forwarding and attendants

The attendant cannot have a Call Forwarding button.

The system does not forward calls to attendants. However the system does forward calls to an attendant group.

Only the attendant, or a phone user with console permission, can activate the Call Forwarding All Calls capability for the following entities:

- A terminating extension group (TEG)
- A direct department calling (DDC) hunt group
- A uniform call distribution (UCD) hunt group
- Data modules

Attendants and users cannot activate or deactivate the Call Forwarding All Calls capability for a vector-controlled split under any circumstances.

Call Forward Busy/Don't Answer

Users use the Call Forward Busy/Don't Answer capability to redirect incoming calls to another destination when the user:

- Is busy on a call
If the user is busy on a call, the system immediately forwards the call. The system does not cause the telephone to ring before the system forwards the call to another destination.
- Does not answer the call within the allowed time interval
If the user does not answer the call, the telephone rings for the allowed time interval. At the end of the interval, the system forwards the call to another destination.

You can restrict access to the Call Forward Busy/Don't Answer capability to specific users.

A user cannot have both the Call Forward Busy/Don't Answer capability and the Call Forwarding All Calls capability active at the same time.

Users activate or deactivate the Call Forward Busy/Don't Answer capability with an FAC or a Call Forward Busy/Don't Answer feature button. Attendants or users that have console permission can also activate or deactivate the feature for another extension with an FAC. Virtual extension users cannot activate or deactivate Call Forwarding Busy/Don't Answer.

Call Forward Busy/Don't Answer cannot be activated for calls to hunt groups, data extensions, a TEG, or an Expert Agent Selection (EAS) agent.

Call Forwarding Off-Net

Users use the Call Forwarding Off Net capability to forward calls to an off-network destination. However, you can restrict access to the Call Forwarding Off Net capability to specific users.

If the Coverage of Calls Redirected Off-Net (CCRON) capability is enabled, and the called user has a coverage path, the system monitors a call for call progress tones. If no one answers the call at the off-network destination, or if the destination is busy, the system returns the call to the internal extension. For more information on the Call Coverage feature, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

The system can bring the call back for call-coverage processing if the principal's coverage criteria are satisfied at the forwarded-to destination.

If the Coverage of Calls Redirected Off Net capability is enabled, but the called user does not have a coverage path, the system does not monitor a call for call progress tones. The system leaves the call at the off-network destination.

When the system redirects an incoming trunk call off the network, the system sets a timer. The timer prevents other incoming trunk calls from redirecting off the network until the timer either expires or is cancelled. The timer prevents calls that are redirected off network from being routed back to the original telephone number from the off-network destination. Calls that are routed back to the original telephone number in this situation effectively create a loop that seizes trunks until trunks are no longer available.

Call Forwarding Override

Users use the Call Forwarding Override capability to call a user, or transfer a call to a user, who has the Call Forwarding feature active. Only the user who answers the forwarded call can use the Call Forwarding Override capability to transfer the call back to the called extension. For example, user A designates user B as the destination for forwarded calls. User B answers a call forwarded from the extension of user A, and uses the Call Forwarding Override capability to transfer the call back to user A.

If you enable the Call Forwarding Override capability on your system, it is available for all users.

Users cannot use the Call Forwarding Override capability to override a call when the system forwards a call:

- To an off-network destination
- From a data user
- From a hunt group

Notifying users when their calls are redirected

You can administer a setting to notify users that they have a capability active that might redirect their calls. For example, if call forwarding is active for a user, you can administer a setting to play a special dial tone when the user goes offhook.

Coverage for unanswered forwarded calls

You can specify that unanswered forwarded calls have call coverage treatment. The system sends an unanswered forwarded call to coverage if you:

- Enable coverage for unanswered forwarded calls for your system
- Enable call forwarding capabilities for the user
- Assign a call coverage path for the user

Security

Users who do not have permission to make calls to an off-network destination cannot use the Call Forwarding feature to forward calls to an off-network destination.

Hardware requirements for Call Forwarding

The Call Forwarding feature requires the following hardware:

- The Call Forwarding Off Net capability requires an available outgoing trunk.

Administering Call Forwarding

The following steps are part of the administration process for the Call Forwarding feature:

- [Viewing the user extensions that have the Call Forwarding capabilities active](#)
- [Assigning the Call Forwarding All Calls capability to a user](#)
- [Removing the Call Forwarding All Calls capability for a user](#)
- [Assigning the Call Forward Busy/Don't Answer capability to a user](#)
- [Removing the Call Forward Busy/Don't Answer capability for a user](#)
- [Assigning the Call Forwarding Off Net capability to a user](#)
- [Removing the Call Forwarding Off Net capability for a user](#)
- [Enabling the Call Forwarding Override capability for your system](#)
- [Disabling the Call Forwarding Override capability for your system](#)

This section describes:

- Any prerequisites for administering the Call Forwarding feature
- The screens that you use to administer the Call Forwarding feature
- Complete administration for the Call Forwarding feature

Prerequisites for administering the Call Forwarding feature

You must complete the following actions before you can administer the Call Forwarding feature:

- Create a class of service (COS) that enables your users to use the capabilities associated with the Call Forwarding feature.
- Ensure that feature access codes (FACs) for Call Forwarding and Call Forward Busy/Don't Answer are available on your system, if you want users to use an FAC for either of the capabilities.
- Enable call coverage for unanswered forwarded calls, if you want the capability on your system.

To create a COS that enables your users to use the capabilities associated with the Call Forwarding feature:

You can create many classes of service. These classes of service enable many features and capabilities, including those for the Call Forwarding feature. You can create a COS that enables none, some, or all of the capabilities associated with the Call Forwarding feature. However, you can assign only one COS to each user. For more information on COS, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

To ensure that an FAC for Call Forwarding All Calls and Call Forward Busy/Don't Answer are available on your system:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the **Feature Access Codes (FAC)** screen ([Figure 102, Feature Access Code \(FAC\) screen](#), on page 467).

Figure 102: Feature Access Code (FAC) screen

FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1 Access Code:	*00	
Abbreviated Dialing List2 Access Code:	*01	
Abbreviated Dialing List3 Access Code:	*02	
Abbreviated Dial - Prgm Group List Access Code:	*03	
Announcement Access Code:	*04	
Answer Back Access Code:	*05	
Auto Alternate Routing (AAR) Access Code:	8	
Auto Route Selection (ARS) - Access Code 1:	*9	Access Code 2: 9
Automatic Callback Activation:	*06	Deactivation: #06
Call Forwarding Activation Busy/DA: *07	All: *08	Deactivation: #09
Call Park Access Code:	*10	
Call Pickup Access Code:	*11	
CAS Remote Hold/Answer Hold-Unhold Access Code:	*50	
CDR Account Code Access Code:	*12	
Change COR Access Code:		
Change Coverage Access Code:	*13	
Contact Closure Open Code:		Close Code:
Contact Closure Pulse Code:		

- 2 If the Call Forwarding Activation Busy/DA and the Call Forwarding Activation All fields each contain an FAC, press **Cancel**.

- 3 If the Call Forwarding Activation Busy/DA field does not contain an FAC, type an FAC in the field.
- 4 If the Call Forwarding Activation All field does not contain an FAC, type an FAC in the field.

For more information, see the “Feature Access Code” feature.
- 5 Press **Enter** to save your changes.

To enable call coverage for unanswered forwarded calls:

- 1 Type **change system-parameters coverage-forwarding**. Press **Enter**.

The system displays the *System Parameters Call Coverage/Call Forwarding* screen ([Figure 103, System Parameters Call Coverage/Call Forwarding screen](#), on page 468).

Figure 103: System Parameters Call Coverage/Call Forwarding screen

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	
Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls: 1	
COVERAGE	
	Keep Held SBA at Coverage Point? y
	External Coverage Treatment for Transferred Incoming Trunk Calls? n
	Immediate Redirection on Receipt of PROGRESS Inband Information? n
	Maintain SBA At Principal? y
	Station Hunt Before Coverage? n
FORWARDING	
	Call Forward Override? y
	Coverage After Forwarding? y

- 2 Perform one of the following actions:
 - If the Coverage After Forwarding field is set to **y**, press **Cancel**.
 - If the Coverage After Forwarding field is set to **n**:
 - Type **y** in the field.
 - Press **Enter** to save your change.

Screens for administering Call Forwarding

Screen Name	Purpose	Fields
<i>Class of Service</i>	Enable Call Forwarding capabilities for users.	• Call Fwd-All Calls • Call Forwarding Busy/DA • Restrict Call Fwd-Off Net
<i>Feature Access Code (FAC)</i>	Assign an FAC for the Call Forwarding capabilities.	• Call Forwarding Activation All • Call Forwarding Activation Busy/DA
<i>Station</i>	Assign a Class of Service (COS) that has the Call Forwarding capabilities enabled.	COS
<i>System-Parameters Coverage/Forwarding</i>	Enable the Call Forwarding Override capability.	• Call Forward Override • Coverage After Forwarding
<i>System-Parameters Customer-Options</i>	Ensure that the system can forward calls to an off-network destination.	Restrict Call Forward Off Net

Viewing the user extensions that have the Call Forwarding capabilities active

To view all the user extensions that have the call forwarding capabilities active:

- 1 Type **list call-forwarding**. Press **Enter**.

This system lists the extensions, with the forwarded destination, that have an active call forwarding capability.

NOTE:

If you have a V1, V2, or V3 system, the **list call-forwarding** command is not available to you. However, you can use the **display station** command to view the call forwarding capabilities that are active for a single user extension.

Assigning the Call Forwarding All Calls capability to a user

To assign the Call Forwarding All Calls capability to a user:

- 1 Type **change station n**, where **n** is the telephone extension number of the user. Press **Enter**.

The system displays the **Station** screen ([Figure 104, Station screen](#), on page 470).

Figure 104: Station screen

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 In the COS field, type the number of a Class of Service that has the Call Forwarding All Calls capability enabled.
- 3 Press **Enter** to save your change.

Removing the Call Forwarding All Calls capability for a user

To remove the Call Forwarding All Calls capability from a user:

- 1 Type **change station *n***, where *n* is the user extension. Press **Enter**.
The system displays the **Station** screen ([Figure 104, Station screen](#), on page 470).
- 2 In the COS field, type the number of a COS that does not have the Call Forwarding All Calls capability enabled.
- 3 Press **Enter** to save your change.

Assigning the Call Forward Busy/Don't Answer capability to a user

Prerequisites

You must complete the following actions before you can assign a COS that has the Call Forward Busy/Don't Answer capability enabled:

- Assign the Call Forward Busy/Don't Answer ring interval for calls that the system forwards to an internal extension.

To assign the Call Forward Busy/Don't Answer ring interval for calls that the system forwards to an internal extension.

- 1 Type **change system-parameters coverage-forwarding**. Press **Enter**.
The system displays the **System Parameters Call Coverage/Call Forwarding** screens ([Figure 105, System Parameters Call Coverage/Call Forwarding screen](#), on page 471).

Figure 105: System Parameters Call Coverage/Call Forwarding screen

```
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
    Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls: 1

COVERAGE
    Keep Held SBA at Coverage Point? y
    External Coverage Treatment for Transferred Incoming Trunk Calls? n
    Immediate Redirection on Receipt of PROGRESS Inband Information? n
    Maintain SBA At Principal? y
    QSIG VALU Coverage Overrides QSIG Diversion with Rerouting? n
    Station Hunt Before Coverage? y

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

- 2 In the Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings) field, type the number of times that a telephone rings before the system forwards the call. The system uses this interval when the Call Forward Busy/Don't Answer capability is active for a user.
- 3 Press **Enter** to save your change.

To assign the Call Forward Busy/Don't Answer capability to a user:

- 1 Type **change station *n***, where *n* is the user telephone extension number. Press **Enter**.
The system displays the **Station** screen ([Figure 104, Station screen](#), on page 470).
- 2 In the COS field, type the number of a COS that has the Call Forward Busy/Don't Answer capability enabled.
- 3 Press **Enter** to save your change.

Removing the Call Forward Busy/Don't Answer capability for a user

To remove the Call Forward Busy/Don't Answer capability for a user:

- 1 Type **change station *n***, where *n* is the user extension. Press **Enter**.
The system displays the **Station** screen ([Figure 104, Station screen](#), on page 470).
- 2 In the COS field, type the number of a COS that does not have the Call Forward Busy/Don't Answer capability enabled.
- 3 Press **Enter** to save your change.

Assigning the Call Forwarding Off Net capability to a user

Prerequisites

You must complete the following actions before you can assign a COS that has the Call Forwarding Off Net capability enabled:

- Ensure that the Call Forwarding Off Net capability is enabled on **Optional Features** screen.
- Ensure that the Call Forwarding Off Net capability is enabled on the **System-Parameters Call Coverage/Call Forwarding** screen.
- Assign a ring interval for the Call Forward Busy/Don't Answer capability for calls that the system forwards to an off-network destination.

To ensure that the Call Forwarding Off Net capability is enabled on the **Optional Features** screen:

- On the **Optional Features** screen, verify that the Restrict Call Forward Off Net field is set to **y**. To view the screen, type **display system-parameters customer-options**. Press **Enter**. If the Restrict Call Forward Off Net field is set to **n**, your system is not enabled for the Call Forwarding Off Net capability. Contact your Avaya representative for assistance before continuing with this procedure.

For a complete description of the System-Parameters Customer-Options screen, [click here](#) or see the *Administrator's Guide for Avaya Communication Manager* for more information.

To ensure that Call Forwarding Off Net capability is enabled on the **System-Parameters Call Coverage/Call Forwarding** screen:

- 1 Type **change system-parameters coverage-forwarding**. Press **Enter**.

The system displays the **System Parameters Call Coverage/Call Forwarding** screens ([Figure 106, System Parameters Call Coverage/Call Forwarding screen](#), on page 472).

Figure 106: System Parameters Call Coverage/Call Forwarding screen

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
Disable call classifier for CCRON over ISDN trunks? n
Disable call classifier for CCRON over SIP trunks? n

- 2 Page through the screens until you see the **Coverage Of Calls Redirected Off-Net Enabled** field.

- If the Coverage Of Calls Redirected Off-Net Enabled field is set to **y**, Press **Cancel**.
- If the Coverage Of Calls Redirected Off-Net Enabled field is set to **n**:
 - Type **y** in the field.
 - Press **Enter** to save your change.

To Assign a ring interval for the Call Forward Busy/Don't Answer capability:

- 1 Type **change system-parameters coverage-forwarding**. Press **Enter**.

The system displays the **System Parameters Call Coverage/Call Forwarding** screens ([Figure 107, System Parameters Call Coverage/Call Forwarding screen](#), on page 473).

Figure 107: System Parameters Call Coverage/Call Forwarding screen

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
Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls:	1
COVERAGE	
Keep Held SBA at Coverage Point?	y
External Coverage Treatment for Transferred Incoming Trunk Calls?	n
Immediate Redirection on Receipt of PROGRESS Inband Information?	n
Maintain SBA At Principal?	y
QSIG VALU Coverage Overrides QSIG Diversion with Rerouting?	n
Station Hunt Before Coverage?	y
FORWARDING	
Call Forward Override?	y
Coverage After Forwarding?	y

- 2 In the Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings) field, type the number of times that a telephone rings before the system forwards the call. The system uses this interval when the Call Forward Busy/Don't Answer capability is active for a user and the forwarded-to number is an off-network destination.
- 3 Press **Enter** to save your change.

To assign the Call Forwarding Off Net capability to a user:

- 1 Type **change station n**, where **n** is the user telephone extension number. Press **Enter**.
The system displays the **Station** screen ([Figure 104, Station screen](#), on page 470).
- 2 In the COS field, type the number of a COS that has the Call Forwarding Off Net capability enabled.
- 3 Press **Enter** to save your change.

Removing the Call Forwarding Off Net capability for a user

To remove the Call Forwarding Off Net capability for a user:

- 1 Type **change station n**, where **n** is the user extension. Press **Enter**.
The system displays the **Station** screen ([Figure 104, Station screen](#), on page 470).
- 2 In the COS field, type the number of a COS that does not have the Call Forwarding Off Net capability enabled.
- 3 Press **Enter** to save your change.

Enabling the Call Forwarding Override capability for your system

To enable the Call Forwarding Override capability for your system:

- 1 Type **change system-parameters coverage-forwarding**. Press **Enter**.

The system displays the *System Parameters Call Coverage/Call Forwarding* screen ([Figure 108, System Parameters Call Coverage/Call Forwarding screen](#), on page 474).

Figure 108: System Parameters Call Coverage/Call Forwarding screen

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	
Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls: 1	
COVERAGE	
	Keep Held SBA at Coverage Point? y
	External Coverage Treatment for Transferred Incoming Trunk Calls? n
	Immediate Redirection on Receipt of PROGRESS Inband Information? n
	Maintain SBA At Principal? y
	Station Hunt Before Coverage? n
FORWARDING	
	Call Forward Override? y
	Coverage After Forwarding? y

- 2 If the Call Forward Override field is set to **y**, press **Cancel**.
- 3 If the Call Forward Override field is set to **n**:
 - Type **y** in the field.
 - Press **Enter** to save your change.

Disabling the Call Forwarding Override capability for your system

To disable the Call Forwarding Override capability for your system:

- 1 Type **change system-parameters coverage-forwarding**. Press **Enter**.

The system displays the *System Parameters Call Coverage/Call Forwarding* screen ([Figure 108, System Parameters Call Coverage/Call Forwarding screen](#), on page 474).

- 2 If the Call Forward Override field is set to **n**, press **Cancel**.
- 3 If the Call Forward Override field is set to **y**:
 - Type **n** in the field.
 - Press **Enter** to save your change.

End-user procedures for Call Forwarding

End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Call Forwarding:

- [Changing the Call Forwarding All Calls destination from an internal telephone](#)
- [Changing the Call Forward Busy/Don't Answer destination from an internal telephone](#)
- [Changing the forwarding destination when a user is at an off-network location](#)
- [Changing the Call Forward Busy/Don't Answer destination when a user is at an off-network location](#)

Changing the Call Forwarding All Calls destination from an internal telephone

To change their call forwarding all calls destination, a user:

- 1 Goes off hook
- 2 Dials the Call Forwarding Activation All feature access code or presses their Call Forwarding Activation All feature button
- 3 Listens for a dial tone
- 4 Dials the extension number of the destination
- 5 Hangs up the telephone after they hear the three-beep tone

Changing the Call Forward Busy/Don't Answer destination from an internal telephone

To change their call forwarding destination, a user:

- 1 Goes offhook
- 2 Dials the Call Forward Busy/Don't Answer activation feature access code or presses their Call Forwarding/Busy Don't Answer feature button
- 3 Listens for a dial tone
- 4 Dials the extension number of the destination
- 5 Hangs up the telephone after they hear the three-beep tone

Changing the forwarding destination when a user is at an off-network location

To change their call forwarding destination, a user:

- 1 Goes offhook
- 2 Dials the telecommuting extension number

- 3 Dials the Extended Call Forward Activate All feature access code or presses their Call Forwarding Activation All feature button
- 4 Listens for a dial tone
- 5 Dials their extension number and presses the # key
- 6 Dials the security code and press the # key
- 7 Listens for a dial tone
- 8 Dials the extension number of the destination

Use no more than 18 digits when you enter your off-network call forwarding destination telephone number. You must include the Trunk Access Code or Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) feature access code among the 18 digits. Do not include the # key that you use to terminate a forwarded-to number among the 18 digits. For more information on the AAR/ARS feature, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

- 9 Hangs up the telephone after they hear the three-beep tone that the system generates to confirm the change

Changing the Call Forward Busy/Don't Answer destination when a user is at an off-network location

To change their call forwarding destination, a user:

- 1 Goes offhook
- 2 Dials the telecommuting extension number
- 3 Dials the Extended Call Forward Activate Busy/Don't Answer feature access code or presses their Call Forward Activate Busy/Don't Answer feature button
- 4 Listens for a dial tone
- 5 Dials their extension number and presses the # key
- 6 Dials the security code and presses the # key
- 7 Listens for a dial tone
- 8 Dials the extension number of the destination

Use no more than 18 digits when you enter your off-network call forwarding destination telephone number. You must include the Trunk Access Code or Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) feature access code among the 18 digits. Do not include the # key that you use to terminate a forwarded-to number among the 18 digits. For more information on the AAR/ARS feature, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

- 9 Hangs up the telephone after they hear the three-beep tone that the system generates to confirm the change

Reports for Call Forwarding

The following reports provide information about the Call Forwarding feature:

- None

Considerations

This section provides information about how the Call Forwarding feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Call Forwarding under all conditions.

- Call classifiers and tone plans for off-network calls

If you send calls off the network 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 downloads the American tone plan regardless of your geographical location.

- Save translation command

If you use the **save translation** command, the software saves the user call forwarding information, including destination information, to tape.

Interactions

- Answer Detection

The Answer Detection feature shares call-classifier resources with the Coverage of Calls Redirected Off-Net capability.

- Attendant Override of Diversion

If an attendant uses the Call Forwarding Override to call a user who has the Call Forwarding feature active, the system sends the call to the user telephone. The system does not forward the call to the forwarded-to destination.

- Automatic Callback and Ringback Queuing

A user cannot activate Automatic Callback if the Call Forwarding feature is active at the called extension. If the user activates Automatic Callback before the Call Forwarding feature is active at the called extension, the system redirects the callback call attempt to the forwarded destination.

- Bridging

The system does not terminate calls to a bridged call appearance when the Call Forward Busy/Don't Answer capability is active at the user extension.

Users cannot bridge onto an off-network call during the time the system is classifying the call.

- **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.

When the Cover All Calls capability is active, and either the Call Forwarding All Calls capability or the Call Forwarding Off Net capabilities is active, the system:

- Forwards incoming priority call
- Redirects all non-priority calls according to the user coverage path
- Does not redirect non-priority calls of the network

- **Call Detail Recording (CDR)**

When the system forwards a call off the network:

- The CDR records the forwarded-from number.
- The system generates a CDR record only after the call is answered at the off-network destination.

If the forced entry of account codes is required, the system does not forward calls to an off-network destination.

- **Call Park**

When a user activates Call Forwarding, and then activates Call Park, the system parks the call at the user extension. The system does not forward the call.

When the system forwards a call, and the forwarded-to extension user parks the call, the system usually parks the call at the forwarded-to extension. The system does not usually park the call at the called extension.

- **Call Pickup/Directed Call Pickup**

If you enable a Temporary Bridged Appearance for the Call Pickup capability, the system maintains a temporary bridged appearance when the forwarded-from user and the forwarded-to user are members of the same call pickup group.

- **Call Prompting**

The Call Prompting feature shares call-classifier resources with the Coverage of Calls Redirected Off-Net capability.

- **Call Visor ASAI**

The Call Visor ASAI feature shares call-classifier resources with the Coverage of Calls Redirected Off-Net capability.

- **Conference**

Users cannot use the Conference feature to add another user onto an off-network call while the system classifies the call.

The system does not classify a call when the system routes a call to an off-network destination in the following circumstance. The system does not classify the off-network call if any of the conference participants is on hold while the conference is initiated. The system does not classify the call even if the Coverage of Calls Redirected Off Net capability is active.

- **Expert Agent Selection (EAS)**
Agents who are logged in at an extension, and who have EAS enabled, cannot activate or deactivate Call Forwarding. If the agent logs out of the extension, the agent can activate or deactivate Call Forwarding for the extension. If the agent logs out of the extension and the agent activates Call Forwarding for the extension, the system forwards calls made to the extension.
- **Hold**
The system does not classify a call when the system routes a forwarded call to the off-network destination in the following circumstance. The system does not classify the off-network call if any party on the call is on hold. The system does not classify the call even if the Coverage of Calls Redirected Off Net capability is active.
- **Intercom-Automatic**
When a user presses an Intercom-Automatic button, and Call Forwarding is active at the user extension that is associated with the button, the system forwards the Intercom-Automatic feature along with the call. However, if the system forwards the call to an off-network destination, the system does not also forward the Intercom-Automatic feature.
- **Interflow**
The system uses the Call Forwarding All Calls capability and the Interflow feature to redirect Automatic Call Distribution (ACD) calls to an ACD split on another system.
- **Intraflow**
The system uses the Call Forwarding feature to route ACD calls from a split to another destination on the same switch.
- **Leave Word Calling (LWC)**
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.
- **Multifrequency Compelled (MFC) Signaling**
MFC Signaling shares call classification resources with the Coverage of Calls Redirected Off-Net capability.
- **Personal Central Office Line (PCOL)**
The system does not forward PCOL calls.
- **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 both Send All Calls and Call Forwarding All Calls are active at an extension, the system:
 - Redirects call to coverage immediately if the system can do so
 - Forwards other calls, such as Priority Calls

If a user has both Send All Calls and Call Forwarding All Calls activate, 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 calls that are ringing on the network. If the caller hangs up, or someone answers the call, the system drops the temporary bridged appearance.

The system does not maintain a temporary bridged appearance when the system forwards calls to an off-network destination.

- **Traffic Reports Removed**

Use the list measurement tone-receiver traffic reports to obtain information on port usage for the Traffic Reports Removed feature.

- **Transfer**

Users cannot transfer a call that the system routes to an off-network destination while the system classifies the call.

Call Park

Use the Call Park feature to retrieve a call that is on hold, from any other telephone within the system. For example, a user can answer a call at one extension, put the call on hold, and then retrieve the call at another extension. Or the user can answer a call at any telephone after an attendant or another user pages the user.

Detailed description of Call Park

This section provides a detailed description of the Call Park feature.

You can set a system-wide expiration interval for parked calls. If no one answers the call before the interval expires, the system redirects the call.

The system redirects a call that is parked to the attendant if the **Deluxe Paging and Call Park Timeout to Originator** field on the **Feature-Related System Parameters** screens is set to **n**. The system redirects a call that is parked to the user who parked the call if the **Deluxe Paging and Call Park Timeout to Originator** field on the **Feature-Related System Parameters screens** is set to **y**.

If you do not administer an attendant or a night service extension, and if you did not administer the Night Service-Trunk Answer from Any Station capability, the system ignores the expiration interval and the call remains parked.

If two parties are connected on a parked call, a third party can create a three-way conference, if the third party answers the call before the interval expires.

The attendant console group can have common shared extensions that the attendant console group uses exclusively for the Call Park feature. The system does not assign the common shared extensions to a telephone. The system stores the common shared extensions in the system translations, and parks calls at the extensions.

The common shared extensions are particularly useful when an attendant pages a user at the request of another user. The attendant parks the calling user on a common shared extension, and announces the extension. The status lamp that is associated with the extension indicates “call parked” or “no call parked,” rather than an active status or an or idle status.

Hardware requirements for Call Park

The Call Park feature requires the following hardware:

- None

Administering Call Park

The following steps are part of the administration process for the Call Park feature:

- [Assigning a call park button to a user of a multiple-call appearance telephone](#)

This section describes:

- Any prerequisites for administering the Call Park feature
- The screens that you use to administer the Call Park feature
- Complete administration for the Call Park feature

Prerequisites for administering the Call Park feature

You must complete the following actions before you can administer the Call Park feature:

- Ensure that feature access codes (FACs) for Call Park are available on your system.
- Administer the Feature-Related System Parameters screens to specify:
 - The minutes that the system leaves a call parked at any extension on your system.
 - That the system routes calls that exceed the call park timeout limit back to the originator of the call.
- Define the common shared extensions on your system.

To ensure that an FAC for Call Park is available on your system:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the *Feature Access Codes (FAC)* screen ([Figure 109, Feature Access Code screen](#), on page 483).

Figure 109: Feature Access Code screen

change feature-access-codes		Page 1 of 8
FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1	Access Code: *00	
Abbreviated Dialing List2	Access Code: *01	
Abbreviated Dialing List3	Access Code: *02	
Abbreviated Dial - Prgm Group List	Access Code: *03	
Announcement	Access Code: *04	
Answer Back	Access Code: *05	
Auto Alternate Routing (AAR)	Access Code: 8	
Auto Route Selection (ARS) -	Access Code 1: *9	Access Code 2: 9
Automatic Callback Activation:	*06	Deactivation: #06
Call Forwarding Activation Busy/DA:	*07 All: *08	Deactivation: #09
Call Park	Access Code: *10	
Call Pickup	Access Code: *11	
CAS Remote Hold/Answer Hold-Unhold	Access Code: *50	
CDR Account Code	Access Code: *12	
Change COR	Access Code:	
Change Coverage	Access Code: *13	
Contact Closure	Open Code:	Close Code:
Contact Closure	Pulse Code:	

2 Perform one of the following actions:

- If the Answer Back Access Code field and the Call Park Access Code field each contain an FAC, your system already has the Feature Access Codes (FACs) that are necessary for the Call Pickup Feature. Press **Cancel**.
- If either the Answer Back Access Code field or the Call Park Access Code field do not contain an FAC, type an FAC in the field. Press **Enter** to save your changes.

For more information on the Feature Access Code feature, including how to change or deactivate an FAC, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

To administer the Feature-Related System Parameters screen for the Call Park feature:

1 Type **change system-parameters features**. Press **Enter**.

The system displays the **Feature-Related System Parameters** screen ([Figure 110, Feature-Related System Parameters screen](#), on page 484).

Figure 110: Feature-Related System Parameters screen

change system-parameters features	Page 1 of 13
FEATURE-RELATED SYSTEM PARAMETERS	
Self Station Display Enabled?	y
Trunk-to-Trunk Transfer:	all
Automatic Callback - No Answer Timeout Interval (rings):	3
Call Park Timeout Interval (minutes):	1
Off-Premises Tone Detect Timeout Interval (seconds):	20
AAR/ARS Dial Tone Required?	y
Music (or Silence) on Transferred Trunk Calls?	no
DID/Tie/ISDN/SIP Intercept Treatment:	attd
Internal Auto-Answer of Attd-Extended/Transferred Calls:	transferred
Automatic Circuit Assurance (ACA) Enabled?	n
Abbreviated Dial Programming by Assigned Lists?	n
Auto Abbreviated/Delayed Transition Interval (rings):	2
Protocol for Caller ID Analog Terminals:	Bellcore
Display Calling Number for Room to Room Caller ID Calls?	n

Figure 111: Feature-Related System Parameters screen

change system-parameters features	Page 3 of 13
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:	5
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:	none
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 Codes Enabled? y
	Authorization Code Length: 7
	Authorization Code Cancellation Symbol: #
	Attendant Time Out Flag? n
	Display Authorization Code? y
Controlled Toll Restriction Replaces:	none

- 2 In the **Call Park Timeout Interval (minutes)** field, type the number of minutes that you want the system to park a call at an extension.
- 3 Press **Next** until you see the **Deluxe Paging and Call Park Timeout to Originator** field ([Figure 111, Feature-Related System Parameters screen](#), on page 484).

- 4 Perform one of the following actions:
 - In the **Deluxe Paging and Call Park Timeout to Originator?** field, type **n** if you want the system to route a parked call, that exceeds the number of minutes that you specified in the **Call Park Timeout Interval (minutes)** field, to the attendant. The system provides **n** as the default entry for this field.
 - In the **Deluxe Paging and Call Park Timeout to Originator?** field, type **y** if you want the system to route a parked call, that exceeds the number of minutes that you specified in the **Call Park Timeout Interval (minutes)** field, to the originator of the call.
- 5 Press **Enter** to save your changes.

To define the common shared extensions on your system:

- 1 Type **change console-parameters**. Press **Enter**.

The system displays the **Console Parameters** screen ([Figure 112, Console Parameters screen](#), on page 485).

Figure 112: Console Parameters screen

change console-parameters	CONSOLE PARAMETERS	Page 2 of 5
TIMING		
Time Reminder on Hold (sec): 30	Return Call Timeout (sec): 30	
Time in Queue Warning (sec):		
INCOMING CALL REMINDERS		
No Answer Timeout (sec):	Alerting (sec):	
Secondary Alert on Held Reminder Calls? y		
ABBREVIATED DIALING		
List1:	List2:	List3:
SAC Notification? n		
COMMON SHARED EXTENSIONS		
Starting Extension:	Count:	

- 2 Page through the screens until you see the **Common Shared Extensions** area.
- 3 In the **Starting Extension** field, type an extension at which you want the attendant to park a call.
- 4 In the **Count** field, type the number of extensions that you want the attendants to have available to park calls. The system uses the information in the **Starting Extension** field and the **Count** field to determine which extensions are available for attendants to park a call. For example, if you type **4300** in the **Starting Extension** field and you type **3** in the **Count** field, the system provides three consecutive extensions, 4300, 4301, and 4302 to park calls.
- 5 Press **Enter** to save your changes.

Screens for administering Call Park

Screen name	Purpose	Fields
<i>Console-Parameters</i>	Specify the extensions where an attendant can park a call.	- Starting Extension - Count
<i>Feature Access Code (FAC)</i>	Define the system-wide FACs to use to park a call and to answer a parked call.	- Answer Back Access Code - Call Park Access Code
<i>Feature-Related System Parameters</i>	Specify the number of minutes that the system parks a call on your system.	Call Park Timeout Interval
	Specify that the system return a call that exceeds the timeout to the originator of the call.	Deluxe Paging and Call Park Timeout to Originator
<i>Station</i>	Assign a call park button to a user with a multiple-call appearance telephone.	Button Assignments

Assigning a call park button to a user of a multiple-call appearance telephone

- 1 Type **change station *n*** where *n* is the telephone number of the extension to which you want to assign a call park button. Press **Enter**.
The system displays the **Station** screen for the extension that you requested ([Figure 113, Station screen](#), on page 487).

Figure 113: Station screen

change station 20001		Page 3 of 4
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:	
3: call-appr	7:	
4: call-park	8:	

- 2 Page through the screens until you see the **Button Assignments** area.
- 3 Type **s** next to the number of the button that you want the user to use to park a call.
- 4 Press **Enter** to save your change.

End-user procedures for Call Park

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Call Park:

- [Using Call Park from a single-line telephone](#)
- [Using Call Park from a multiple-call appearance telephone](#)
- [Using Call Park from an attendant console](#)
- [Retrieving a parked call](#)

Using Call Park from a single-line telephone

To use the Call Park feature from a single-line telephone:

- Flash the switch hook.
- Dial the call park feature access code (FAC).
- Hang up.

Using Call Park from a multiple-call appearance telephone

To use the Call Park feature from a multiple-call appearance telephone:

- Press the transfer button or the conference button.
- Dial the call park access code FAC.
- Press the transfer button or the conference button.

Using Call Park from an attendant console

To use the Call Park feature from an attendant console:

- Press the start button.
- Dial the call park access code FAC.
- Dial the extension where the attendant wants to park the call.
- Press the release button.

An attendant can also use the Direct Extension Selection with Busy Lamp Field capability with the Call Park feature. For more information on the Direct Extension Selection with Busy Lamp feature, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Retrieving a parked call

To use the Call Park feature to retrieve a call, perform one of the following actions:

- Press the same call park button that was used to park the call.
- Dial the answer back FAC.

Reports for Call Park

The following reports provide information about the Call Park feature:

- None

Considerations for Call Park

This section provides information about how the Call Park feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Call Park under all conditions. The following considerations apply to Call Park:

- A User can park only one call at an extension at one time, even if the extension has multiple call appearances. A user can park a conference calls that has two to five participants. A user cannot park a conference call that has six participants. The sixth position must remain open so someone can retrieve the call.
- Neither a user nor an attendant can park a call on a group extension. If a group member parks a call, the system parks the call at the extension of the group member. Group members can belong to the following groups:
 - A coverage answer group
 - A uniform call distribution (UCD) hunt group
 - A direct department calling (DDC) hunt group
 - A terminating extension group (TEG)
- If all appearances on a parked telephone are busy and no attendant or night-service extensions are configured when the call park timeout expires, the system:
 - Drops the call if a coverage path does not exist.
 - Does not drop the call if a coverage path exists.

Interactions for Call Park

This section provides information about how the Call Park feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Call Park in any feature configuration.

- Abbreviated Dialing
A user presses the Abbreviated Dialing button to park calls, or retrieve calls that are parked.
- Attendant Console
Assign the common shared extensions to the optional Attendant Selector Console in the 00 through 09 block of numbers, on the bottom row, in any hundreds group so that the attendant can easily identify the extensions. The lamp that is associated with the number indicates “call parked” or “no call parked,” rather than a busy status or an idle status. For more information on the Attendant features, [click here](#), or see the *Administrator’s Guide for Avaya Communication Manager*.
- Automatic Wakeup
Neither a user nor an attendant can park Automatic Wakeup calls.
- Bridged Call Appearance
If a user, that is active on a bridged call appearance, activates Call Park, the system parks the call on the primary extension associated with the bridged call appearance.

- **Call Vectoring**
 - Neither a user nor an attendant can park a call on a vector directory number (VDN) extension.
 - Neither a user nor an attendant can park a call that is undergoing vector processing.
- **Code Calling Access**

The system automatically parks a user or an attendant on the extension of the party that is paged when the user or attendant:

 - Is using the Paging feature.
 - Dials the Code Calling Access feature access code (FAC)
 - Dials the extension of the party that is paged
- **Conference**

Both users and attendants can park Conference calls.
- **Data Privacy and Data Restriction**

The system automatically deactivates the Data Privacy feature and the Data Restriction feature when a user or attendant parks a call.
- **Drop**

If a user receives an external call, and the user pushes the **drop** button after the user parks the call, the call is no longer parked.

If a user receives an internal call, and the user pushes the **drop** button after the user parks the call, the call remains parked. The system drops the call only when the user who parks the call hangs up.
- **Loudspeaker Paging Access**

Neither a user nor an attendant can park calls to paging zones.
- **Music-on-Hold**

If a parked call involves only one party, the user that is parked hears music-on-hold. The parking user also hears music after the user parks the call and the system generates a 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.
- **Tenant Partitioning**

If an attendant parks a call on a common shared extension, and tenant partitioning is *not* active, the system routes the call to the attendant group when the call exceeds the call park timeout.

If an attendant parks a call on a shared extension, and tenant partitioning is active, the system routes the call to the attendant who parked the call when the call exceeds the park timeout interval.

The system functions as described in the preceding circumstances regardless of whether the **Deluxe Paging and Call Park Timeout to Originator** field of the **Feature-Related System-Parameters** screen is set to y or n.
- **Transfer**

If the **Transfer Upon Hang-up** field on the **Feature Related System-Parameters** screen is set to y, a user does not need to press the Transfer button a second time to park a call.

Call Pickup

Use the Call Pickup feature to allow users to answer call for one another. The feature requires that the users be members of the same pickup group. With the related Directed Call Pickup capability, users can answer calls for one another, even though the users are not members of the same call pickup group.

Call Pickup supports the following capabilities:

- **Directed Call Pickup**
Users use Directed Call Pickup to answer the calls of any other user in the system. These users can be members of different call pickup groups, or might not be a member of any call pickup group.
- **Extended Group Pickup**
Use Extended Group Pickup to combine call pickup groups into extended call pickup groups, and to combine extended call pickup groups into other extended call pickup groups.

Detailed Description

This section provides a detailed description of the Call Pickup feature.

Call Pickup

Users use Call Pickup to answer calls for one another. The users must be in the same call pickup group. A call pickup group is comprised of users that you assign to a group with a unique, identifying number. A user can be a member of only one call pickup group. However, call pickup groups can also be combined into extended call pickup groups.

Call ringing and status lamp flashing

The members of a call pickup group know that another group member is receiving a call in two ways:

- The group members can hear the other telephone ring.
- The status lamps of all the group members flash.

If a call pickup group relies on ringing to know when another group member receives a call, the group members should be located near each other. The members should be located near each other, so that they can hear the ringing of the other telephones.

When a group member users Call Pickup to answer a ringing call, the telephone of the called member stops ringing.

When the group members see their status lamps flashing, and use Call Pickup to answer the call:

- The status lamp of the answering group member lights, but does not flash, for the duration of the call.
- The telephone of the called group member stops ringing.
- The status lamp and the call appearance on the telephone of the called group member continue to flash for the duration of the call.
- The status lamps of the other group members go out.

Note that call pickup alerting causes only a status lamp that is not already lit to flash. For example, when a group member receives a call, the status lamps of those group members that are not talking on a call answered with Call Pickup do not flash. These lamps do not flash because these lamps are steadily lit while the members are talking on the call.

The system uses an algorithm to select the call when multiple calls ring or alert in a call pickup group. The system searches the extensions of the call pickup group until the system finds an extension with a call that is eligible to be answered with Call Pickup. The system selects this call to be answered. The next time that a group member answers a call with Call Pickup, the system bypasses the extension that was answered most recently, and starts the search at the next extension. For example, if a group member attempts to use Call Pickup when two calls are ringing at extension A, and one call is ringing at extension B, the system selects the calls in the following order:

- 1 One of the calls to extension A
- 2 The call to extension B
- 3 The remaining call to extension A

The system also determines which call that a group member answers when multiple calls ring or alert at the same telephone. The system selects the call with the lowest call appearance, which is usually the call appearance that is nearest to the top of the telephone. For example, when calls ring or alert at the second and the third call appearances, the system selects the call on the second call appearance for the user to answer.

Directed Call Pickup

Users can use the Directed Call Pickup capability to answer the calls of any other user on the system. The system does not require that the users be members of the same call pickup group before users can answer calls for one another.

Extended Group Pickup

With the Extended Group Pickup capability, you can combine:

- Pickup groups into extended pickup groups. This capability is called *simple* extended group pickup.
- Extended pickup groups into other extended pickup groups. This is called *flexible* extended group pickup.

Users cannot use a call pickup button with the Extended Group Pickup capability.

Hardware requirements for Call Pickup

The Call Pickup feature requires the following hardware:

- None

Administering Call Pickup

The following steps are part of the administration process for the Call Pickup feature:

- Administering a call pickup button
 - [Assigning a call pickup button to a user telephone](#)
 - [Changing a call pickup button for a user telephone](#)
 - [Removing a call pickup button from a user telephone](#)
- Administering a call pickup group
 - [Assigning a user to a call pickup group](#)
 - [Removing a user from a call pickup group](#)
- Administering the Directed Call Pickup capability
 - [Assigning the Directed Call Pickup capability to a user](#)
 - [Removing the Directed Call Pickup capability from a user](#)
- Administering an extended call pickup group for a call pickup group
 - [Assigning a pickup group to an extended pickup group](#)
 - [Removing a pickup group from an extended pickup group](#)
- Administering an extended call pickup group for an extended call pickup group
 - [Assigning extended pickup groups to another extended pickup group](#)
 - [Removing an extended pickup group from an extended pickup group](#)

Screens for administering Call Pickup

Screen Name	Purpose	Fields
<i>Class of Restriction</i>	Create a Class of Restriction (COR) for a user to use the Directed Call Pickup capability.	• Can Be Picked Up By Directed Call Pickup? • Can Use Directed Call Pickup?
<i>Extended Pickup Group</i>	Combine pickup groups into extended pickup groups.	Pickup Group Number
	Combine extended pickup groups into another extended pickup group.	Pickup Group Number

Screen Name	Purpose	Fields
Feature-Related System Parameters	Enable the Directed Call Pickup capability.	Directed Call Pickup
	Enable the Extended Group Pickup capability.	Extended Group Call Pickup
Pickup Group	Assign a user to a pickup group.	- Group Name - Extension
Station	Assign the call pickup button to a user extension.	Button Assignments
	Assign a COR to a user extension.	COR

Assigning a call pickup button to a user telephone

Prerequisites

You must complete the following actions before you assign a call pickup button to a user telephone:

- Ensure that the Call Pickup Access feature access code (FAC) is available on your system.

To ensure that the Call Pickup Access feature access code (FAC) is available on your system:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the **Feature Access Code (FAC)** screen [Figure 114, Feature Access Code screen](#), on page 494.

Figure 114: Feature Access Code screen

FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1 Access Code:	*00	
Abbreviated Dialing List2 Access Code:	*01	
Abbreviated Dialing List3 Access Code:	*02	
Abbreviated Dial - Prgm Group List Access Code:	*03	
Announcement Access Code:	*04	
Answer Back Access Code:	*05	
Auto Alternate Routing (AAR) Access Code:	8	
Auto Route Selection (ARS) - Access Code 1:	*9	Access Code 2: 9
Automatic Callback Activation:	*06	Deactivation: #06
Call Forwarding Activation Busy/DA: *07	All: *08	Deactivation: #09
Call Park Access Code:	*10	
Call Pickup Access Code:	*11	
CAS Remote Hold/Answer Hold-Unhold Access Code:	*50	
CDR Account Code Access Code:	*12	
Change COR Access Code:		
Change Coverage Access Code:	*13	
Contact Closure Open Code:		Close Code:
Contact Closure Pulse Code:		

2 Page through the screens until you see the Call Pickup Access Code field.

3 Perform one of the following actions:

- If this field contains a FAC, press **Cancel**.
- If this field does not contain an FAC:
 - Type a FAC in the field.
 - Press **Enter** to save your change.

For more information on the Feature Access Code feature, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

To assign a call pickup button to a user telephone:

1 Type **change station *n***, where *n* is the telephone number of the extension to which you want to assign a call pickup button. Press **Enter**.

The system displays the **Station** screen that you requested ([Figure 115, Station screen](#), on page 495).

Figure 115: Station screen

STATION		
SITE DATA Room: Jack: Cable: Floor: Building:	Headset? n Speaker? n Mounting: d Cord Length: 0 Set Color:	
ABBREVIATED DIALING List1: List2: List3:		
BUTTON ASSIGNMENTS 1: call-appr 2: call-appr 3: call-appr 4: 5: 6: 7: 8: 9: call-pkup 10: dn-dst 11: goto-cover 12: send-calls Ext:		

2 Page through the screens until you find the BUTTON ASSIGNMENTS area.

3 Move to the button number that you want to use for call pickup. You can use any of the buttons.

4 Type **call-pkup** after the button number.

5 Press **Enter** to save your changes.

Changing a call pickup button for a user telephone

To change a call pickup button on a user telephone:

- 1 Type **change station *n***, where *n* is the telephone number of the extension for which you want to change a call pickup button. Press **Enter**.
The system displays the **Station** screen that you requested [Figure 115, Station screen](#), on page 495.
- 2 Page through the screens until you find the BUTTON ASSIGNMENTS area.
- 3 Move to the existing **call-pkup** button.
- 4 Press **Clear**.
- 5 Move to the button number that you want to use for call pickup.
- 6 Type **call-pkup** after the button number.
- 7 Press **Enter** to save your changes.

Removing a call pickup button from a user telephone

To remove a call pickup button from a user telephone:

- 1 Type **change station *n***, where *n* is the telephone number of the extension for which you want to remove a call pickup button. Press **Enter**.
The system displays the **Station** screen that you requested ([Figure 115, Station screen](#), on page 495).
- 2 Page through the screens until you find the BUTTON ASSIGNMENTS area.
- 3 Move to the existing **call-pkup** button.
- 4 Press **Clear**.
- 5 Press **Enter** to save your changes.

Assigning a user to a call pickup group

To assign a user to a call pickup group, perform one of the following steps:

- If the pickup group to which you want to assign the user does *not* exist:

- 1 Type **add pickup-group next**. Press **Enter**.

The system assigns a number to the call pickup group and the system displays a **Pickup Group** screen ([Figure 116, Pickup Group screen](#), on page 497).

Figure 116: Pickup Group screen

PICKUP GROUP			
Group Number: 1			
Group Name:			
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:			

2 Type the extension of each group member. You can add a maximum of 50 extensions.

3 Press **Enter** to save your changes.

The system automatically completes the Name fields when you press **Enter**.

- If the pickup group to which you want to assign the user exists:

1 Type **change pickup-group *n***, where *n* is the number of the pickup group to which you want to assign the user. Press **Enter**.

The system displays the **Pickup Group** screen ([Figure 117, Pickup Group screen](#), on page 497).

Figure 117: Pickup Group screen

PICKUP GROUP			
Group Number: 1			
Group Name:			
GROUP MEMBER ASSIGNMENTS			
Ext	Name (first 26 characters)	Ext	Name (first 26 characters)
1: 31101	1b1801 x31101 cabinet num	14:	
2: 32101	2b0501 x32101 cabinet num	15:	
3:		16:	
4: 34101	4a1001 x34101 cabinet num	17:	
5:		18:	
6:		19:	
7:		20:	
8:		21:	
9:		22:	
10:		23:	
11:		24:	
12:		25:	
13:			

- 2 Type the extension of the user to whom you want to assign to the pickup group.
- 3 Press **Enter** to save your changes.

The system automatically completes the Name fields when you press **Enter**.

Removing a user from a call pickup group

To remove a user from a call pickup group:

- 1 Type **change pickup-group *n***, where *n* is the number of the pickup group from which you want to remove the user. Press **Enter**.

The system displays the **Pickup Group** screen ([Figure 117, Pickup Group screen](#), on page 497).

- 2 Move to the extension that you want to remove.
- 3 Press **Clear**.
- 4 Press **Enter** to save your changes.

Assigning the Directed Call Pickup capability to a user

Prerequisites

You must complete the following actions before you can assign the Directed Call Pickup capability to a user:

- Ensure that the Directed Call Pickup capability is available on your system.
- Ensure that a Class of Restriction (COR) exists for a user to use the Directed Call Pickup capability.

To ensure that the Directed Call Pickup capability is available on your system:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the **Feature-Related System Parameters** screen ([Figure 118, Feature-Related System Parameters screen](#), on page 499).

Figure 118: Feature-Related System Parameters screen

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:	5
Call Pickup Alerting?	n
Temporary Bridged Appearance on Call Pickup?	y
Call Pickup on Intercom Calls?	y
Directed Call Pickup?	y
Extended Group Call Pickup:	none
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 Codes Enabled? y
	Authorization Code Length: 7
	Authorization Code Cancellation Symbol: #
	Attendant Time Out Flag? n
	Display Authorization Code? y
Controlled Toll Restriction Replaces:	none

2 Page through the screens until you see the Directed Call Pickup field.

3 Perform one of the following actions:

- If the Directed Call Pickup field is set to **y**, press **Cancel**.
- If the Directed Call Pickup field is set to **n**:
 - Type **y** in the field.
 - Press **Enter** to save your changes.

To ensure that a COR exists for a user to use the Directed Call Pickup capability:

1 Type **change cor *n***, where ***n*** is the number of the COR that you want to verify. Press **Enter**.

The system displays the ***Class of Restriction*** screen that you requested ([Figure 119, Class of Restriction screen](#), on page 500).

Figure 119: Class of Restriction screen

CLASS OF RESTRICTION	
COR Number: 1	
COR Description:	
FRL: 7	APLT? y
Can Be Service Observed? y	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? n
Restriction Override: all	Facility Access Trunk Test? n
Restricted Call List? n	Can Change Coverage? n
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 Phone)? y
Can Be Picked Up By Directed Call Pickup? y	Can Use Directed Call Pickup? y
	Group Controlled Restriction: inactive

In the Can Be Picked Up By Directed Call Pickup field, type y.

- 2 Page through the screens until you see the **Can Be Picked Up By Directed Call Pickup?** field.
- 3
- 4 In the Can Use Directed Call Pickup field, type y.
- 5 Press **Enter** to save your changes.

For more information on the Class of Restriction (COR) feature, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

To assign Directed Call Pickup permission to a user, you must complete the following procedure:

- 1 Type **change station n**, where **n** is the telephone number of the user to whom you want to assign Directed Call Pickup permission. Press **Enter**.

The system displays the **Station** screen for the extension that you requested ([Figure 120, Station screen](#), on page 501).

Figure 120: Station screen

STATION		
Extension: 30019	Lock Messages? n	BCC: 0
Type: 4612	Security Code: *	TN: 1
Port: S04007	Coverage Path 1:	COR: 1
Name: station 30019	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 30019	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english		
	Media Complex Ext:	
	IP SoftPhone? n	

- 2 Move to the COR field. Type the number of the COR that you want to assign to the user that has Directed Call Pickup permissions.
- 3 Press **Enter** to save your changes.

Removing the Directed Call Pickup capability from a user

To remove the Directed Call Pickup capability from a user:

- 1 Type **change station *n***, where *n* is the telephone number of the extension from which you want to remove a COR that supports the Directed Call Pickup capability. Press **Enter**.
The system displays the **Station** screen for the extension that you requested ([Figure 120, Station screen](#), on page 501).
- 2 Move to the COR field. Type the number of a COR that you want to assign to the user that does not have Directed Call Pickup permissions.
- 3 Press **Enter** to save your changes.

Assigning a pickup group to an extended pickup group

Prerequisites

You must complete the following actions in the order listed before you assign a pickup group to an extended pickup group:

- Ensure that the Extended Group Pickup capability is available on your system.
- Ensure that the Extended Group Call Pickup Access feature access code (FAC) is available on your system.

To ensure that the Extended Group Pickup capability is available on your system:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the *Feature-Related System Parameters* screen ([Figure 121, Feature-Related System Parameters screen](#), on page 502).

Figure 121: Feature-Related System Parameters screen

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:	5
Call Pickup Alerting?	n
Temporary Bridged Appearance on Call Pickup?	y
Call Pickup on Intercom Calls?	y
Directed Call Pickup?	y
Extended Group Call Pickup:	simple
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 Codes Enabled? y
	Authorization Code Length: 7
	Authorization Code Cancellation Symbol: #
	Attendant Time Out Flag? n
	Display Authorization Code? y
	Controlled Toll Restriction Replaces: none

-
- 2 Page through the screens until you see the Extended Group Call Pickup field.
 - 3 Perform one of the following actions:
 - If the Extended Group Call Pickup field contains the word **simple**, press **Cancel**.
 - If the Extended Group Call Pickup field does *not* contain the word **simple**:
 - Type **simple** in the field.
 - Press **Enter** to save your changes.

To ensure that the Extended Group Call Pickup Access FAC is available on your system:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the *Feature Access Code (FAC)* screen ([Figure 122, Feature Access Code \(FAC\) screen](#), on page 503).

Figure 122: Feature Access Code (FAC) screen

FEATURE ACCESS CODE (FAC)			
Data Origination Access Code:	*17		
Data Privacy Access Code:	*18		
Directed Call Pickup Access Code:	*19		
Emergency Access to Attendant Access Code:	*20		
Enhanced EC500 Activation:		Deactivation:	
Extended Call Fwd Activate Busy D/A *21 All:	*22	Deactivation:	#22
Extended Group Call Pickup Access Code:	*39		
Facility Test Calls Access Code:	*24		
Flash Access Code:	*25		
Group Control Restrict Activation:	*26	Deactivation:	#26
Hunt Group Busy Activation:	*27	Deactivation:	#27
ISDN Access Code:			
Last Number Dialed Access Code:	*29		
Leave Word Calling Message Retrieval Lock:	*30		
Leave Word Calling Message Retrieval Unlock:	*31		
Leave Word Calling Send A Message:	*32		
Leave Word Calling Cancel A Message:	*33		
Malicious Call Trace Activation:		Deactivation:	
Meet-me Conference Access Code Change:			

2 Page through the screens until you see the Extended Group Call Pickup Access Code field.

3 Perform one of the following actions:

- If the Extended Group Call Pickup Access Code field contains a FAC, press **Cancel**.
- If the Extended Group Call Pickup Access Code field does not contain a FAC:
 - Type a FAC in the field.
 - Press **Enter** to save your changes.

For more information on the Feature Access Code (FAC) feature, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

To assign pickup groups to an extended pickup group:

1 Type **change extended-pickup-group *n***, where *n* is the number of the extended pickup group to which you want to assign the pickup group. Press **Enter**.

The system displays an **Extended Pickup Group** screen ([Figure 123, Extended Pickup Group screen](#), on page 504).

Figure 123: Extended Pickup Group screen

EXTENDED PICKUP GROUP			
Extended Group Number: 1			
Pickup Number	Pickup Group Number	Pickup Number	Pickup Group Number
0:	1	13:	
1:	12	14:	
2:	15	15:	
3:		16:	
4:		17:	
5:		18:	
6:		19:	
7:		20:	
8:		21:	
9:		22:	
10:		23:	
11:		24:	
12:			

- 2 Type the numbers of the extended pickup groups that you want to assign to this extended pickup group.
- 3 Press **Enter** to save your changes.

Removing a pickup group from an extended pickup group

To remove a pickup group from an extended pickup group:

- 1 Type **change extended-pickup-group *n***, where *n* is the number of the extended pickup group from which you want to remove the pickup group. Press **Enter**.
The system displays the **Extended Pickup Group** screen ([Figure 123, Extended Pickup Group screen](#), on page 504).
- 2 Move to the extended pickup group number that you want to remove from the pickup group.
- 3 Press **Clear**.
- 4 Press **Enter** to save your changes.

Assigning extended pickup groups to another extended pickup group

Prerequisites

You must complete the following actions in the order listed before you assign extended pickup groups to another extended pickup group:

- Ensure that the Extended Group Pickup capability is available on your system.
- Ensure that the Extended Group Call Pickup Access FAC is available on your system.

To ensure that the Extended Group Pickup capability is available on your system:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the *Feature-Related System Parameters* screen ([Figure 124, Feature-Related System Parameters screen](#), on page 505).

Figure 124: Feature-Related System Parameters screen

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:	5
Call Pickup Alerting?	n
Temporary Bridged Appearance on Call Pickup?	y
Call Pickup on Intercom Calls?	y
Directed Call Pickup?	y
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 Codes Enabled? y
	Authorization Code Length: 7
	Authorization Code Cancellation Symbol: #
	Attendant Time Out Flag? n
	Display Authorization Code? y
Controlled Toll Restriction Replaces:	none

-
- 2 Page through the screens until you see the Extended Group Call Pickup field.
 - 3 Perform one of the following actions:
 - If the Extended Group Call Pickup field contains the word **flexible**, press **Cancel**.
 - If the Extended Group Call Pickup field does not contain the word **flexible**:
 - Type **flexible** in the field.
 - Press **Enter** to save your changes.

To ensure that the Extended Group Call Pickup Access FAC is available on your system:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the *Feature Access Code (FAC)* screen ([Figure 125, Feature Access Code screen](#), on page 506).

Figure 125: Feature Access Code screen

FEATURE ACCESS CODE (FAC)	
Data Origination Access Code: *17	
Data Privacy Access Code: *18	
Directed Call Pickup Access Code: *19	
Emergency Access to Attendant Access Code: *20	
Enhanced EC500 Activation: *21	Deactivation: #22
Extended Call Fwd Activate Busy D/A *21 All: *22	Deactivation: #22
Extended Group Call Pickup Access Code: *39	
Facility Test Calls Access Code: *24	
Flash Access Code: *25	
Group Control Restrict Activation: *26	Deactivation: #26
Hunt Group Busy Activation: *27	Deactivation: #27
ISDN Access Code: *29	
Last Number Dialed Access Code: *30	
Leave Word Calling Message Retrieval Lock: *31	
Leave Word Calling Message Retrieval Unlock: *32	
Leave Word Calling Send A Message: *33	
Leave Word Calling Cancel A Message: *33	
Malicious Call Trace Activation: *33	Deactivation: #27
Meet-me Conference Access Code Change: *33	

2 Page through the screens until you see the Extended Group Call Pickup Access Code field.

3 Perform one of the following actions:

- If the Extended Group Call Pickup Access Code field contains a FAC, press **Cancel**.
- If the Extended Group Call Pickup Access Code field does *not* contain a FAC:
 - Type a FAC in the field.
 - Press **Enter** to save your changes.

For more information on the Feature Access Code (FAC) feature, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

To assign an extended pickup group to another extended pickup group:

1 Type **change extended-pickup-group n**, where **n** is the number of the extended pickup group to which you want to assign another extended pickup group. Press **Enter**.

The system displays an **Extended Pickup Group** screen ([Figure 126, Extended Pickup Group screen](#), on page 507).

Figure 126: Extended Pickup Group screen

EXTENDED PICKUP GROUP			
Extended Group Number: 1			
Pickup Number	Pickup Group Number	Pickup Number	Pickup Group Number
0:	1	13:	
1:	12	14:	
2:	15	15:	
3:		16:	
4:		17:	
5:		18:	
6:		19:	
7:		20:	
8:		21:	
9:		22:	
10:		23:	
11:		24:	
12:			

- 2 Type the extended pickup group number that you want to add to this extended pickup group.
- 3 Press **Enter** to save your changes.

Removing an extended pickup group from an extended pickup group

To remove an extended pickup group from an extended pickup group:

- 1 Type **change extended-pickup-group *n***, where *n* is the extended pickup group from which you want to remove an extended pickup group. Press **Enter**.
The system displays the *Extended Pickup Group* screen ([Figure 126, Extended Pickup Group screen](#), on page 507).
- 2 Move to the extended pickup group number from which you want to remove the extended pickup group.
- 3 Press **Clear**.
- 4 Press **Enter** to save your changes.

End-user procedures for Call Pickup

End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Call Pickup:

- Answering a call with Call Pickup
To pick up the call of another user, a user goes offhook. The user then either presses the call pickup button, or enters the call pickup feature access code (FAC).
- Answering a call with Extended Group Pickup
Each extended pickup group, whether comprised of pickup groups or other extended pickup groups, has a unique identifying number. When users use the Extended Group Pickup capability to answer a call, the users enter the FAC for Extended Group Call Pickup, and then the unique extended pickup group number.

Reports for Call Pickup

The following reports provide information about the Call Pickup feature:

- None

Considerations for Call Pickup

This section provides information about how the Call Pickup feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Call Pickup under all conditions.

- Exclusion is not supported for call pickup calls.
- A member of a call pickup group who makes a call to another group member cannot use Call Pickup to answer the call.

Interactions for Call Pickup

This section provides information about how the Call Pickup feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Call Pickup in any feature configuration.

- Abbreviated Dialing
A user can store either:
 - Both the feature access code (FAC) for Directed Call Pickup and a telephone number in a single Abbreviated Dial button
 - Only the Directed Call Pickup FAC in an Abbreviated Dial button

- **Attendant**

An attendant can use the Directed Call Pickup capability to answer calls. However, other users cannot use the Directed Call Pickup capability to answer a call that alerts at an attendant console.
- **Automatic Callback and Ringback Queuing**

Neither the call pickup group members nor the Directed Call Pickup users can answer a Callback calls
- **Bridged Call Appearance**
 - If call pickup alerting is active, and a bridged call appearance rings on the telephone of a call pickup group member, other group members cannot answer the call.
 - If the call pickup alerting capability is not active, and a bridged call appearance rings at the telephone of a call pickup group member, other group members can answer the call.
 - If the temporary bridged appearance capability on the Call Pickup feature is enabled, a Temporary Bridged Appearance is maintained at the called telephone. The called party can then bridge onto the call after another call pickup group member answers the call.
 - You cannot use the Directed Call Pickup capability to answer a call that alerts at a bridged call appearance.
- **Call Coverage**

If a user has a call-coverage temporary bridged appearance, the user can use Directed Call Pickup to answer a redirected call that alerts at the telephone of another covering user.
- **Call Detail Recording (CDR)**

CDR records the extension number that the caller dials.
- **Call Forwarding**

If the Temporary Bridged Appearance capability on the Call Pickup feature is enabled, the system maintains a temporary bridged appearance if the forwarded-to extension belongs to the same call pickup group as the forwarded-from extension. If the Temporary Bridged Appearance capability on the Call Pickup feature is not enabled, the system does not maintain a temporary bridged appearance.
- **Call Pickup Alerting**

When a pickup group member uses the Directed Call Pickup capability to answer the ringing call of another group member, and the call is the only call that is ringing for any member of the pickup group, the call pickup alerting lamp goes out.
- **Call Waiting**

A user cannot use the Call Pickup capability to answer a Call Waiting call.
- **Conference**

If call pickup alerting is enabled, and a user uses the Conference feature after the user answers the call, the call pickup status lamp goes out. If additional calls come into the pickup group, the status lamp of the user flashes.
- **Consult**

If the Temporary Bridged Appearance capability is not enabled for the Call Pickup feature, the system presents a Consult call from the covering user as an idle call appearance.

- Expert Agent Selection (EAS)

EAS agents use the Directed Call Pickup capability to:

- Have other agents answer their calls
- Answer the calls of other agents

The Class of Restriction (COR) of the agent overrides the COR of the extension where the agent is logged in.

If an agent is logged in to a telephone extension, and if the Directed Call Pickup capability is active for both the agent and the extension, a user can answer a call in two ways. The user can use either the agent Login ID or the telephone extension to answer the call with Directed Call Pickup.

Call Waiting Termination

Use the Call Waiting Termination feature to automatically notify a user with a single-line telephone, who is active on a call, that a second call is waiting.

Detailed description of Call Waiting Termination

This section provides a detailed description of the Call Waiting Termination feature.

Users with a single-line telephone can place a call on hold to answer a waiting call. After a user answers the call that is waiting, the user can return to the call that is held, or the user can toggle back and forth between the two calls. A user with a single-line telephone can connect to only one call at a time.

When a call is waiting for a user, the user hears:

- One quick burst of tone when a call from another user is waiting
- Two quick bursts of tone when an outside call, or a call that is handled by an attendant, is waiting
- Three quick bursts of tone when a priority call is waiting

Note that the system does not support special ring tones 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, if an attendant handles the call, the user hears a busy tone, unless the Attendant Call Waiting Indication field on the **Station** screen is set to **y**.

You assign Call Waiting Termination on a per-telephone basis. For a virtual extension, you assign Call Waiting Termination on the physical station.

Hardware requirements for Call Waiting Termination

The Call Waiting Termination feature requires the following hardware:

- A single-line telephone

Administering Call Waiting Termination

The following steps are part of the administration process for the Call Waiting Termination feature:

- [Assigning Call Waiting Termination](#)

This section describes:

- Any prerequisites for administering the Call Waiting Termination feature
- The screens that you use to administer the Call Waiting Termination feature
- Complete administration procedures for the Call Waiting Termination feature

Prerequisites for administering Call Waiting Termination

You must complete the following actions before you can administer the Call Waiting Termination feature:

- Administer the Feature-Related System Parameters screen to support the Call Waiting Termination feature on your system.

To administer the Feature-Related System Parameters screen to support the Call Waiting Termination feature on your system:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the *Feature-Related System Parameters* screen ([Figure 127, Feature-Related System Parameters screen](#), on page 512).

Figure 127: Feature-Related System Parameters screen

change system-parameters features	Page 10 of 14
FEATURE-RELATED SYSTEM PARAMETERS	
Pull Transfer: y	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): 4
Vector Disconnect Timer (min):	Network Feedback During Tone Detection? y
Hear Zip Tone Following VOA? y	System Updates Time On Station Displays? y
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	
Onhook Dialing on 607/2420/4600/6400/8400 Terminals? y	
ITALIAN DCS PROTOCOL	
Italian Protocol Enabled? n	

- 2 Page through the screens until you see the Repetitive Call Waiting Tone field.
- 3 In the Repetitive Call Waiting Tone field, perform one of the following actions:
 - If you want the users and the attendants to hear a repetitive call waiting tone when they use the Call Waiting Termination feature, type **y**.
 - If you do not want the users and the attendants to hear a repetitive call waiting tone when users and attendants use the Call Waiting Termination feature, type **n**.
- 4 In the Repetitive Call Waiting Interval (sec) field, type the number of seconds between the repetitive call waiting tones.
- 5 Press **Enter** to save your changes.

Screens for administering Call Waiting Termination

Screen name	Purpose	Fields
<i>Feature-Related System Parameters</i>	Specify that a user or an attendant hears a repetitive tone when a call is waiting.	• Repetitive Call Waiting Tone?
	Specify the number of seconds between each repetitive call waiting tone.	• Repetitive Call Waiting Interval (sec)
<i>Station</i>	Enable the Call Waiting Termination feature for a user.	• Call Waiting Indication
	Enable the Call Waiting Termination feature for an attendant.	• Att. Call Waiting Indication

Assigning Call Waiting Termination

To assign the Call Waiting feature to a user or an attendant:

- 1 Type **change station n**, where **n** is the telephone number of the extension to which you want to assign the Call Waiting Termination feature. Press **Enter**.

The system displays the **Station** screen for the extension that you requested ([Figure 128, Station screen](#), on page 514).

Figure 128: Station screen

change station 20001		Page 2 of 3
FEATURE OPTIONS		STATION
LWC Reception: spe		Coverage Msg Retrieval? y
LWC Activation? y		Auto Answer: none
LWC Log External Calls? n		Data Restriction? n
CDR Privacy? n		Call Waiting Indication? y
Redirect Notification? y		Att. Call Waiting Indication? y
Per Button Ring Control? n		Distinctive Audible Alert? y
Bridged Call Alerting? n		Adjunct Supervision? y
Switchhook Flash? y		
Ignore Rotary Digits? n		
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
Service Link Mode: as-needed		
Multimedia Mode: basic		
MWI Served User Type:		
AUDIX Name:		
		Coverage After Forwarding? s
		Direct IP-IP Audio Connections? y
Emergency Location Ext: 103		IP Audio Hairpinning? y
Precedence Call Waiting? y		

2 Page through the screens until you see the Call Waiting Indication field.

3 In the Call Waiting Indication field, perform one of the following actions:

- If you want to activate Call Waiting Termination for the user, type **y**.

If you set the Call Waiting Indication field to **y**, calls that a user or an attendant originate, and calls that originate from the outside, wait at the single-line telephone if the telephone is busy. The system sends a distinctive call-waiting tone to the user.

The system denies the Call Waiting Termination feature to a user if the:

- Data Restriction field on the **Station** screen is set to **y**.
- Switchhook Flash field on the **Station** screen is set to **n**.
- Class of Service that you assign to the user activates the Data Privacy feature for the user.

- If you want to deactivate Call Waiting Termination for the user, type **n**.

4 In the Att. Call Waiting Indication field, perform one of the following actions:

- Type **y** to activate Attendant Call Waiting Termination for the user.

If you set the Att. Call Waiting Indication field to **y**, calls that an attendant originates, or that an attendant extends, wait at the single-line telephone if the telephone is busy. The system sends a distinctive call-waiting tone to the user.

The system denies the Call Waiting Termination feature to a user if the:

- Data Restriction field on the **Station** screen is set to **y**.
- Switchhook Flash field on the **Station** screen is set to **n**.
- Class of Service (COS) that you assign to the user activates the Data Privacy feature for the user.

- Type **n** to deactivate Call Waiting Termination for the user.

5 Press **Enter** to save your change.

Reports for Call Waiting Termination

The following reports provide information about the Call Waiting Termination feature:

- None

Considerations for Call Waiting Termination

This section provides information about how the Call Waiting Termination feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Call Waiting Termination under all conditions. The following considerations apply to Call Waiting Termination:

- Call Waiting is only available for users who have single-line telephones. Calls to multiple-appearance telephones do not wait, because the system routes these calls to an idle call appearance.
- 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 you activate Call Waiting for a user of an analog single-line telephone, and the user starts a conference call, the system denies the Call Waiting feature to the user while the user is on the conference call.

Interactions for Call Waiting Termination

This section provides information about how the Call Waiting Termination feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Call Waiting Termination in any feature configuration.

- The system denies the use of the Call Waiting Termination feature when any of the following conditions or features are active at a single-line telephone:
 - Another Call Waiting call
 - Automatic Callback (to or from the telephone)
 - Data Privacy
 - Data Restriction
- Call Pickup and Directed Call Pickup

A member of a call pickup group cannot use the Call Pickup feature or the Directed Call Pickup capability to pick up a Call Waiting call.

Caller ID

Use the Caller ID feature to collect and display the calling party name and the calling party number, which the central office (CO) sends on incoming calls. Caller ID is also known as Incoming Call Line Identification (ICLID).

Caller ID supports the following capabilities:

- Caller ID (ICLID) on analog trunks
- Caller ID (ICLID) on digital trunks

Detailed description of Caller ID

This section provides a detailed description of the Caller ID feature.

Communication Manager stores and displays up to 15 characters of Incoming Call Line Identification (ICLID) information, which the central office (CO) sends on incoming calls. If the information is longer than 15 characters, the software truncates the information to 15 characters. If the caller ID information is not received, the system displays the trunk group name and the trunk access code (TAC).

In the US, the CO sends both calling party name and calling party number, if this information is available. In Japan the CO sends only the calling party number. This information is sent on a CO loop-start trunk in the U.S. In Japan, this information is sent on either a CO loop-start trunk, a Direct Inward Dialing (DID) trunk, or a Direct Inward and Outward Dialing (DIOD) trunk.

Caller ID (ICLID) on analog trunks

Caller ID on analog trunks is also known as Bellcore Calling Name ID. This capability allows the system to accept calling name information from a local exchange carrier (LEC) network that supports the Bellcore calling name specification. The system can also send calling name information in the correct format if Bellcore calling name ID is properly administered. The following caller ID protocols are supported:

- Bellcore (default). US protocol (Bellcore transmission protocol with 212 modem protocol)
- V23-Bell. Bahrain protocol (Bellcore transmission protocol with V.23 modem protocol).

Caller ID (ICLID) on digital trunks

In the US, a central office can send the calling party name and the calling party number over digital trunks for display on digital telephones. The display of calling party name and calling party number works with all Communication Manager Digital Communications Protocol (DCP) and Basic Rate Interface (BRI) digital telephones that have either a 40-character or a 32-character alphanumeric display.

Hardware requirements for Caller ID

The Caller ID feature requires the following hardware:

- A TN429D CO Trunk circuit pack, or later configuration
- For ICLID on digital trunks, Caller ID requires a digital telephone that has either a 40-character or 32-character alphanumeric display.
- For ICLID on analog trunks, Communication Manager supports the 7315H System 25 telephones or the 7317H series System 25 telephones.

Administering Caller ID

The following steps are part of the administration process for the Caller ID feature:

- [Displaying ICLID Information](#)

This section describes:

- Any prerequisites for administering the Caller ID feature
- The screens that you use to administer the Caller ID feature
- Complete administration procedures for the Caller ID feature

Prerequisites for administering Caller ID

You must complete the following actions before you can administer the Caller ID feature:

- View the **Optional Features** screen, and ensure that the **G3 Version** field is set to **V6** or later, and that the **Analog Trunk Incoming Call ID** field is set to **y**. If the **G3 Version** field is not set to **V6** or later, or if the **Analog Trunk Incoming Call ID** field is set to **n**, your software is not enabled for the Caller ID feature. Contact your Avaya representative for assistance.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

Screens for administering Caller ID

Screen name	Purpose	Fields
Optional Features	Ensure that the Analog Trunk Incoming Call ID field is set to y .	Analog Trunk Incoming Call ID

Screen name	Purpose	Fields
<i>Trunk Group</i>	Set this field to 120 for ICLID.	Incoming seizure (msec)
	Set up the trunk group to receive ICLID information.	Receive Analog Incoming Call ID
	Set to incoming or two-way .	Direction

Displaying ICLID Information

To set up a trunk group to receive calling party information and display the calling party name and the calling party number on the telephone display of the called party.

- 1 Type **change trunk group *n***, where *n* is the number of the trunk group for which you want to set up ICLID. If you want to set up a new trunk group, type **add trunk group next**.

The system displays the *Trunk Group* screen ([Figure 129, Trunk Group screen](#), on page 519).

Figure 129: Trunk Group screen

change trunk-group 5 Page 1 of x

TRUNK GROUP

Group Number: ____
 Group Name: _____
 Direction: _____
 Dial Access? ____
 Queue Length: ____
 Comm Type: _____
 BCC: ____

Group Type: co____
 COR: ____
 Outgoing Display? ____
 Busy Threshold: ____
 Auth Code? ____
 Trunk Flash? ____
 ITC? ____

CDR Reports: ____
 TN: ____ TAC: ____
 Trunk Signaling Type: ____
 Night Service: ____
 Incoming Destination: _____

TRUNK PARAMETERS

Trunk Type (in/out): _____
 Outgoing Dial Type: _____
 Digit Treatment: _____
 Analog Loss Group: ____
 Incoming Dial Tone? ____
 Incoming Calling Number - Delete: ____
 Bit Rate: _____
 Disconnect Supervision - In? ____ Out? ____
 Answer Supervision Timeout: ____

Incoming Rotary Timeout(sec): ____
 Incoming Dial Type: _____
 Disconnect Timing(msec): ____
 Digits: _____
 Sig Bit Inversion: none
 Digital Loss Group: ____
 Insert: _____
 Synchronization: _____
 Duplex: ____

Receive Answer Supervision? ____

- 2 In the Group Type field, type **co**, **did**, or **diod**.
- 3 In the Direction field, type **incoming** or **two-way**.

NOTE:

When the Group Type is **diod**, the Direction field defaults to **two-way**. When the Group Type is **did**, the Direction field is hidden, because all calls are incoming.

- 4 Click **Next** until the system displays the *Trunk Group* screen *Trunk Features* page. ([Figure 130, Trunk Group Trunk Features screen](#), on page 520).

Figure 130: Trunk Group Trunk Features screen

change trunk-group 5	TRUNK GROUP	Page 2 of x
TRUNK FEATURES		
ACA Assignment? n	Measured: none	Maintenance Tests? y
Suppress # Outpulsing? n	Data Restriction? n	
Incoming Tone (DTMF) ANI: no	Receive Analog Incoming Call ID: Bellcore	

- 5 In the Receive Analog Incoming Call ID field, type **Bellcore** for the U.S., or **NTT** for Japan.
- 6 Click **Next** until the system displays the *Trunk Group* screen *Administrable Timers* page. ([Figure 131, Trunk Group Administrable Timers screen](#), on page 520).

Figure 131: Trunk Group Administrable Timers screen

change trunk-group 5	TRUNK GROUP	Page 3 of x
ADMINISTRABLE TIMERS		
Incoming Disconnect(msec): 500		
Incoming Dial Guard(msec): 70		
Flash Length(msec): 540	Incoming Seizure(msec): 120	
END TO END SIGNALING	Incoming Incomplete Dial Alarm (sec): 255	
Tone (msec): 350	Pause (msec): 150	

- 7 In the Incoming Seizure (msec) field, type **120**. The correct setting for ICLID is **120**.
- 8 Press **Enter** to save your changes.

Reports for Caller ID

The following reports provide information about the Caller ID feature:

- None

Considerations for Caller ID

This section provides information about how the Caller ID feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Caller ID under all conditions. The following considerations apply to Caller ID:

- If you use Incoming Caller ID (ICLID) on analog trunks that are connected to a Direct Inward and Outward Dialing (DIOD) central office (CO) trunk circuit pack, do not put these trunks in an outgoing Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS) route pattern. The loop-start trunks that the DIOD CO trunk circuit pack supports do not provide answer supervision. Thus the potential for toll fraud exists.

Interactions for Caller ID

This section provides information about how the Caller ID feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Caller ID in any feature configuration.

- **Attendant Display Features**
A call that is redirected to either the attendant or the attendant queue causes the display on the station of the attendant to match that of the station display of the connected party.
- **Automatic Display of Incoming Call Identification**
If a new call comes in while the station user is off-hook and connected to a call, the display automatically shows the new incoming call identity for 30 seconds. After 30 seconds, the display returns to the identity of the original call. If the system redirects the call after a few rings, the display returns to identity of the original call. If the system shows a new incoming call, and then that call drops, the display returns to the identity of the original call.
- **Bridged Call**
The system shows incoming call identity on both the primary station and the bridged station.
- **Call Forwarding**
 - Forwarded-From Station Display. The system does not show any information on the station of the called principal.
 - Forwarded-To Station Display. The system shows the identity of the calling party and the called party and the reason (R) code. If the forwarded-to station is on a different switch, the called party information does not forward.
- **Call Pickup**
 - Called Party Station Display. Shows the identity of the calling party.
 - Answering Party Station Display. If Call Pick-Up answers an ICLID call, the system shows the identities of both the calling party and the called party.

- **Call Coverage**
 - **Called Party Display.** The display of the called party shows the identity of the calling party until the coverage party answers the call. If the coverage party answers the call, the station display of the called party goes blank. If the called party temporarily bridges in after the coverage party answers the call, the displays of the coverage party and the called party change to indicate a conference call.
 - **Coverage User Station Display.** The station display of the coverage user shows the same display as the station display of the connected party.
- **Call Vector Routing**

When an ICLID call coming from analog trunks transfers to a Vector Directory Number (VDN), the incoming calling number is directed to that VDN so call vector routing can be based on the ICLID information.

The Automatic Number Identification (ANI) that is received for the incoming call, by way of inband signaling or ISDN, forwards with a route-to step over a trunk that supports inband or ISDN ANI delivery.
- **Distributed Communications System (DCS)**

If Communication Manager has both DCS and Integrated Services Digital Network (ISDN) displays, the system shows the ICLID information in DCS formats.
- **Hold**

When Hold is activated, the display becomes blank. The party who activates the Hold then reads the identity of the newly connected party. The display of the held station remains unchanged. When the party deactivates the Hold, the system refreshes the display to indicate the current state of the call.
- **Malicious Call Trace (MCT)**

When MCT is activated for a particular call, the system displays incoming calling numbers to controller stations.
- **Tandem Operations**

The system passes the calling party name and the calling party number to the terminating server over ISDN trunks with DCS+.
- **Transfer**

When the system transfers an ICLID call, the display of the transferred-from station becomes blank. The display of the transferred-to station shows the identity of the transferred-from party if the transfer is not yet completed. Once the transfer is complete, the transferred-to station shows the identity of the calling party.

Centralized Attendant Service

Use the Centralized Attendant Service (CAS) feature to allow attendants within a private network to serve all branch locations from a central or a main location.

Centralized Attendant Service (CAS) supports the following capabilities:

- Branch-generated call identification tones

Detailed description of Centralized Attendant Service

This section provides a detailed description of the Centralized Attendant Service (CAS) feature.

With Centralized Attendant Service (CAS), you can provide attendant services in a private network from a central location. Each branch in a centralized attendant service has its own listed directory number (LDN) or other type of access from the public network. With this feature, the system routes incoming calls to a branch, and calls that users make directly to the attendants, to the centralized attendant. CAS uses release link trunks (RLT) to direct calls.

The CAS main system operates independently of the individual CAS branch systems. The operation for CAS main system traffic is identical to the operation of a stand-alone system.

Each branch in a CAS network connects to the main office through RLTs. These trunks provide paths for:

- Sending incoming attendant-seeking trunk calls at the branch to the main location for processing and extending the calls back to the branch. Both parts of a call use the same trunk.
- Returning timed-out waiting and held calls from the branch to the main location
- Routing calls from the branch to the main location

Two queues are associated with CAS calls. One queue is at the main office, and the other queue is at the branch. If idle RLTs are available from the branch to the main location, RLTs are seized. CAS calls are then queued at the main location, along with other attendant-seeking calls. If all RLTs are in use, CAS calls to the attendant are queued at the branch in an RLT queue. The length of the queue can vary from 1 to 100 calls. You set the length of the queue when you administer the RLT group.

Backup service provides for all CAS calls to be sent to a backup extension in the branch if all RLTs:

- Are maintenance-busy
- Are out of service
- If the attendant presses a Backup button that is not lit.

To activate this feature and to provide notification that backup service is in effect, assign the backup extension to a Backup button and an associated status lamp.

- The status lamp remains lit as long as backup service is in effect.
- To deactivate the CAS feature, the attendant presses the Backup button while the status lamp is lit. The system does not send calls to the backup extension unless all RLTs are maintenance-busy, or out of service.

The attendant can put a CAS call from a branch on Remote Hold. The branch holds the call, and drops the RLT. After a timeout, which is the same as the timed reminder for an attendant-held call, the branch automatically attempts to route the call back to the attendant. The returning call can queue for the RLT. Attendants can use Remote Hold when the attendant must put a call on hold to prevent the unnecessary use of RLTs.

Branch-generated call identification tones

The branch in a CAS network generates call identification tones, and transmits the tones to the CAS attendant through RLTs. These tones indicate the type of call from the branch, or the status of a call extended to or held at the branch. The attendant hears these tones in the console handset before the attendant is connected to the caller.

The following identification tones might vary by country.

- Incoming trunk calls: 480 Hz (100 ms), 440 Hz (100 ms), 480 Hz (100 ms) in sequence. The attendant hears this tone immediately after the attendant lifts the handset.
- Calls from a branch telephone to the main attendant, or calls that are transferred by a branch telephone to the main attendant: 440 Hz (100 ms), silence (100 ms), 440 Hz (100 ms) in sequence. The attendant hears this tone immediately after the attendant lifts the handset.
- Calls that the system extends to an idle telephone, or recall on Does Not Answer: The attendant hears ringback tone (300 ms), then connection to the normal ringing cycle.
- Calls that the system extends to a busy telephone, automatically waiting, or recall on Attendant Call Waiting: 440 Hz (100 ms).
- Calls that the system extends to a busy telephone, waiting denied, or not provided: A busy tone.
- Remote Hold or Remote Hold recall: A series of four to six cycles of 440 Hz (50 ms), and then silence (50 ms).
- Recall on Does Not Answer: A burst of ringback tones (300 ms), then connection to normal ringback at any point in the cycle.
- Recall from a call that is on Remote Hold: A series of four to six cycles of 440 Hz (50 ms), and then silence (50 ms).
- Recall from a call that is waiting at a single line telephone: A burst of 440 Hz (100 ms).

The centralized attendant at the main location has access, through RLTs, to all outgoing trunk facilities at the branches in a CAS network. To extend an incoming LDN call to an outgoing trunk at a branch, an attendant can dial the access code, and then allow the caller to dial the rest of the number. The attendant can also dial the complete outgoing number.

Calls that are extended to busy single-line telephones at the branch wait automatically. If a call is in queue, the user hears a busy signal. When Station Hunting and Send All Calls is administered, the system routes the call along the administered path. If any extended call is waiting, and is unanswered within an administered interval, the branch system returns the call to the attendant. Call Waiting does not apply to multiappearance telephones. If no appearances are available, busy tone is sent to the attendant, who tells the caller that the line is busy.

The system also routes calls from telephones at the branch to an attendant over RLTs that are seized by the branch system. To reach the attendant, a branch caller dials the attendant-group access code. This access code is administrable. The default is 0. The conversation between the branch caller and the attendant ties up the seized RLT, but calls of this type are usually short.

Hardware requirements for Centralized Attendant Service

The Centralized Attendant Service (CAS) feature requires the following hardware:

- An attendant console
- To set up RLTs, a port is required on a TN722B, a TN767, or a TN464D circuit pack. A TN722B or a TN767 circuit pack provides 24 ports. A TN464D circuit pack provides 24 or 32 ports.

Administering Centralized Attendant Service

This section contains prerequisites and the screens for administering the Centralized Attendant Service (CAS) feature:

Prerequisites for administering Centralized Attendant Service

You must complete the following actions before you can administer the Centralized Attendant Service (CAS) feature:

- Set up the attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.
- On the **Optional Features** screen, ensure that the Async. Transfer Mode (ATM) PNC field is set to **y**. If this field is set to **n**, contact your Avaya representative before you continue with this procedure.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

Screens for administering Centralized Attendant Service

Screen name	Purpose	Fields
Optional Features	Enable port network connectivity (PNC).	Async. Transfer Mode (ATM) PNC
Attendant Console	Assign feature buttons.	Any available button field in the FEATURE BUTTON ASSIGNMENTS area.

Screen name	Purpose	Fields
<i>Console-Parameters</i>	Set up parameters for CAS.	• CAS • RLT Trunk Group Number • Timed Reminder on Hold • Return Call Timeout (sec)
<i>Station</i>	On a multiappearance telephone, assign feature buttons.	Any available button field in the FEATURE BUTTON ASSIGNMENTS area.
<i>Trunk Group</i>	Set up a release link trunk (RLT) group.	All
<i>Feature Access Code (FAC)</i>	Set up a feature access code (FAC) for CAS Remote Hold.	CAS Remote Hold/ Answer Hold-Unhold Access Code

For more information, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Reports for Centralized Attendant Service

The following reports provide information about the Centralized Attendant Service (CAS) feature:

- None

Considerations for Centralized Attendant Service

This section provides information about how the Centralized Attendant Service (CAS) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Centralized Attendant Service (CAS) under all conditions. The following considerations apply to Centralized Attendant Service (CAS):

- CAS reduces the number of attendants that are required at a branch. For example, a chain of department stores can have a centralized attendant at the main store to handle calls for all of the branch stores.
- In a CAS network, systems can function either as branches or as the main location. A branch can connect to only one main location.
- A branch can have an attendant. Access to the branch attendant must be by way of an individual attendant extension. Incoming trunk calls in a CAS network can bypass branch attendants. The centralized attendant can route these incoming trunk calls back to the branch attendant.

- Branch calls terminate on the CAS main system based day-destination or night-service destination of the incoming RLT trunk group. An attendant console might not always answer or extend incoming CAS calls.

If someone other than an attendant answers a CAS call, that person can either press the **Flash** button on a multiappearance telephone, or flash the switch hook on a single-line telephone, to extend the call back to the branch. The branch reaction to flash signals and the branch application of tones is the same whether an attendant, or someone other than an attendant, answers or extends the call.

- If an extended call returns unanswered to the main attendant, the called party at the branch does not drop. The called party at the branch continues to be alerted until the caller releases. This process allows the attendant to talk to the caller, and then extend the call again, if the caller wants, without redialing the number.
- If an extended CAS call recall times out and goes to coverage, but is unanswered, then the branch leaves the extended-to party ringing. The system drops coverage.
- When an analog telephone call goes to coverage, the telephone drops from the call. This process is the exception to the branch leaving the extended-to party ringing. If the main attendant extends a call to an analog telephone, and that call goes to coverage and later returns to the main attendant, the call is treated as an incoming LDN call. If the user requests, the attendant must reextend the call.
- On an incoming CAS call to the main attendant, the system displays the **Name** field from the **Trunk Group** screen for that RLT to the attendant. Therefore, you must administer the **Name** field on the **Trunk Group** screen to provide meaningful branch identification information.
- The Music-on-Hold feature at a branch applies to two stages of LDN calls: during call extension, and when the call is on Remote Hold.

Interactions for Centralized Attendant Service

This section provides information about how the Centralized Attendant Service (CAS) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Centralized Attendant Service (CAS) in any feature configuration.

- **Abbreviated Dialing**
The main attendant can use an Abbreviated Dialing button to extend CAS calls after the attendant obtains branch dial tone.
- **Attendant Auto-Manual Splitting**
The Split lamp and button do not function on CAS calls to the main location that are extended through RLTs. Attendant conference does not function on CAS calls.
- **Attendant Control of Trunk Group Access**
If a branch attendant has control of an outgoing RLT trunk group, the system routes new attendant-seeking calls to the branch attendant.
- **Attendant Override of Diversion**
Use Attendant Override of Diversion with CAS.
- **Attendant Serial Calling**
Attendant Serial Calling does not work for CAS calls.

- Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS)
You can use AAR and ARS to route CAS calls.
- Automatic Circuit Assurance (ACA)
When CAS is activated, the ACA **Referral Calls** field on the ***Feature-Related System Parameters*** screen must be set to **local**. The system interprets a referral destination of 0 as the local attendant, if a local attendant exists. The CAS attendant cannot activate or deactivate ACA referral calls at a branch location.
- Busy-Indicator Buttons
Busy indicators can identify incoming calls over an RLT. You can also use Busy indicators to dial after the attendant starts to extend a call.
- Call Coverage
Use Call Coverage to redirect calls to a centralized attendant. Do not redirect calls to a CAS backup extension for backup service through Send All Calls to the coverage path of the backup extension.
- Call Detail Recording (CDR)
If the CAS main RLT has the CDR option selected, the system generates CDR records for incoming CAS calls.
- Call Forwarding
Do not forward calls to a CAS extension.
- Call Park
If a CAS attendant parks a call and the call returns to the attendant after the Call Park expiration interval, the attendant hears incoming trunk-call notification.
- Class of Restriction (COR)
Since COR information is not passed over RLTs, fully restricted service allows all CAS calls. Therefore, CAS allows the system to complete a public network call to a fully-restricted telephone.
- Distributed Communications System (DCS) operation
If an RLT trunk group is administered as a DCS trunk, the DCS message displays instead of the name of the incoming RLT trunk group on an incoming CAS call to the attendant. When the attendant answers the call, the attendant hears call identification tones. These tones indicate that the call is a CAS call. The attendant must use a TRUNK-NAME button to obtain the name of the RLT trunk group.
- DXS and DTGS Buttons
DXS and DTGS buttons at the main attendant console can be used with CAS. With the DXS button, the attendant hears ringback tone after a delay of a few seconds.
- Emergency Access to the Attendant
For CAS Branch Emergency Access calls that generate by a feature access code, the system routes Off-Hook Alert to the branch attendant group. If there is no attendant in the branch, the system routes the call to the administered Emergency Access Redirection Extension of the branch. When the branch system is in CAS Backup Service, the system routes the calls to the backup telephone. The call is treated as a normal call.

- **Hunt Groups**

If the system directs an incoming CAS call to a hunt group, the system does not redirect the call to the coverage path of the hunt group. Depending on the circumstances, the attendant can get a busy tone or ringing.
- **Last Number Dialed (LND)**

An attendant cannot extend calls with the LND feature.
- **Leave Word Calling (LWC)**

If a message is left for a branch user, and the attendant at the CAS system tries to retrieve the message, permission is denied.
- **Night Service, Night Console Service**

When the attendant places the CAS main location in Night Service, CAS calls terminate at the CAS main night-service destination. When an attendant places a branch location in Night Service, CAS calls route to the branch night console, to the LDN night telephone, or to the TAAS telephone.
- **Night Service, Trunk Answer from Any Station (TAAS)**

In a multisystem DCS environment with CAS, transferring incoming trunk calls through Night Service Extension or TAAS varies. Such a transfer depends on the:

 - Home system of the transferred-to telephone
 - Home system of the connected trunk
 - Type of night-service function that is chosen. The type of night-service function can be Night Service Extension, TAAS, or both.
- **Extending a CAS call by a nonattendant**

CAS branch calls terminate at the CAS main location, based on the day-destination or night-service destination of the incoming RLT trunk group. You can also answer a CAS call with the TAAS feature.

Usually, a nonattendant presses the Flash button to extend a CAS call. However, if the nonattendant does not have a Flash button, a nonattendant can extend the call in one of the following ways:

 - Multiappearance telephone users can press the Conference or Transfer button, and then dial the extension. To complete the call, the user drops the call. To drop the extended-to party, the user presses the Conference or Transfer button again.
 - Single line telephone users can flash the switch hook, and then dial the extension. To complete the call, the user drops the call. To drop the extended-to party, the user flashes the switch hook again.
- **Holding a CAS call by a nonattendant**

A nonattendant with a multiappearance telephone can press the Hold button to hold a CAS call.
- **Releasing a CAS call by a nonattendant**

A nonattendant can drop the RLT by going on-hook, then use the Disconnect or Drop button. A nonattendant can also drop the RLT by selecting another call appearance.
- **Nonattendant. Display Trunk Name**

If a nonattendant with a display telephone presses the Trunk Name button while the nonattendant is active on a trunk call, the system displays the value in the NAME field from the **Trunk Group** screen.

- Security Violation Notification

CAS attendants cannot receive referral calls from branch locations.

- Timed Reminder

You can set the timer value for recalling held calls at the attendant console on the **Console** screen.

If an attendant at the CAS main location transfers a call from a branch to an extension at the main location, the timed reminder does not apply. The call does not return to the attendant if unanswered. If a branch call is unanswered, the branch timed reminder times out, and the system routes the call to a new RLT trunk, and back to a CAS main attendant.

- Trunk-Name button

Use the Trunk-Name button when you make an outgoing call over a trunk that is administered to have no outgoing display.

Class of Restriction

Use the Class of Restriction (COR) feature to:

- Define different levels of call origination and termination privileges
- Apply administration settings to all objects that share the same COR number
- Identify what CORs can be service observed and what CORs can be a service observer

Detailed description of Class of Restriction

This section provides a detailed description of the Class of Restriction feature.

You can use CORs to restrict communication between point A and point B. For example, a user tries to establish a communication path between point A and point B. The system checks whether the CORs have permission to communicate with one another. If the CORs have permission, the system completes the call. If the CORs do not have permission, the system does not complete the call. You control the level of restriction that the COR provides.

CORs have other applications as well. You can apply administration settings to a COR, then assign that COR to objects (facilities) in the system. This use of CORs makes it easier to administer functions across a wide range of objects. CORs are assigned to a variety of objects. Objects can be:

- Telephones
- Trunks
- Agent loginIDs
- Data modules

Finally, you can set up CORs that are service observing and service observed. You can assign a COR to be a service observer. Then you identify what other CORs that the user can observe. You can also set up a COR to be serviced observed.

To set up a COR, you select a COR number, from 0 to 95. You then assign a name for the COR that clearly reflects the purpose or members of the COR. You then use the Class of Restriction screens to select what restrictions, if any, apply to the COR. After you set up a COR, you assign the COR number to objects on your system.

Many objects can share the same COR number. You must administer a COR for the following objects:

- Agent loginIDs
- Access endpoints
- Announcements and audio sources
- Attendant console
- Authorization codes
- Console parameters
- Hunt groups

Class of Restriction

Detailed description of Class of Restriction

- Loudspeaker paging
- Data modules
- Remote access (each barrier code has a COR)
- Telephones
- Terminating Extension Group (TEG)
- Trunk groups
- Vector Directory Numbers (VDN)

Strategy for assigning CORs

When you administer your system, the best strategy is to assign CORs to similar groups or objects. For example, you might create a unique COR for each type of user or facility, such as:

- Call center agents
- Account executives
- Supervisors
- Administrative assistants
- Paging zones
- Data modules

You might also want to create a unique COR for each type of restriction.

You can assign the same COR to more than one object. Objects with the same COR might be similar objects, such as two telephones, or different objects, such as a telephone and a trunk group.

To enhance your system security, you can:

- Assign a separate COR to incoming trunk groups and outgoing trunk groups, and then restrict calls between the two groups.
- Set appropriate calling party restrictions and Facility Restriction Levels (FRLs) to limit the calling permissions as much as possible.

Types of Restrictions

Calling party restrictions

Calling party restrictions define the privileges for telephones that make outbound calls. If you do not need to restrict telephones that make outbound calls, assign a COR with the **Calling Party Restriction** field set to **none**.

You can use calling party restrictions for:

- Unrestricted telephones
- Trunk groups
- Terminating Extension Groups (TEG)
- Uniform Call Distribution (UCD) groups

- Direct Department Calling (DDC) groups
- Data modules
- Attendant groups
- Individual attendant extensions

All-Toll restrictions and TAC-Toll restrictions

Toll restrictions prevent users from placing public-network calls to certain toll-call numbers. You assign toll restrictions to outgoing trunk groups on the **Trunk Group** screen. You disable Trunk Access Code (TAC)-toll restrictions for specific outgoing trunk groups on the **Trunk Group** screen.

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.

Calls coming into a trunk are denied if:

- The **outward** restriction is applied to the **Calling Party Restriction** field on the **Class of Restriction** screen
- The calls use the Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS) feature

Called party restrictions

Called party restrictions define the privileges for telephones that receive inbound calls. If you do not need to restrict telephones that receive inbound calls, assign a COR with the **Called Party Restriction** field set to **none**.

You can use calling party restrictions for:

- Unrestricted telephones
- Trunk groups
- Terminating Extension Groups (TEG)
- Uniform Call Distribution (UCD) groups
- Direct Department Calling (DDC) groups
- Data modules
- Attendant groups
- Individual attendant extensions

Even if the system redirects a call from one telephone to another, the system checks for called party restrictions only at the called telephone. For example, if a called telephone with no restrictions activates the Call Forwarding feature to a restricted telephone, the system still completes the call.

Class of Restriction

Detailed description of Class of Restriction

Inward restrictions

You can use inward restrictions to allow users to receive only internal calls. With inward restrictions, users at assigned telephones cannot receive:

- public network calls
- attendant originated calls
- attendant extended calls

The system checks only the COR of the originally called telephone, unless you administer a three-way COR on conference calls and transfer calls. The system routes denied calls to one of the following:

- intercept tone
- a recorded announcement
- 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 calls that an attendant extends. The system can redirect calls to a telephone with manual terminating line restrictions. The system checks only the COR of the originally called telephone.

The system routes the following calls to the attendant:

- local Central Office (CO) calls
- foreign exchange (FX) calls

The system redirects Direct Inward Dialing (DID) calls to one of the following:

- intercept tone
- a recorded announcement
- the attendant

Public restrictions

Public restrictions prohibit users from receiving public network calls. The system routes denied calls to:

- intercept tone
- a recorded announcement
- the attendant

Public restrictions still allow users to receive internal calls from other telephones, or calls that are extended from the attendant.

Termination restrictions

You can use termination restrictions to prohibit users from receiving any calls. These users can still originate calls. The system routes DID or Advanced Private-Line Termination calls to a recorded announcement or to the attendant.

Fully restricted service

With fully restricted service, users cannot make or receive public network calls. Users who have fully restricted service cannot use authorization codes to deactivate this feature.

COR-to-COR restrictions

You can restrict calls from one COR to another COR. Use COR-to-COR calling restrictions to prohibit access to users to specific telephones. You can use COR-to-COR restrictions to prohibit access to users to specific trunk groups, such as central office (CO) trunk groups. Any or all trunk groups can be in a trunk restriction COR. The system routes restricted calls to intercept tone.

COR-to-COR calling permissions are on page 3 of the *Class of Restriction* screens.

Hardware requirements for Class of Restriction

The Class of Restriction feature requires the following hardware:

- None

Administering Class of Restriction

The following steps are part of the administration process for the Class of Restriction (COR) feature:

- [Displaying administered CORs](#)
- [Setting up a COR](#)
- [Allowing users to change their own COR](#)

This section describes:

- Any prerequisites for administering the Class of Restriction feature
- The screens that you use to administer the Class of Restriction feature
- Complete administration procedures for the Class of Restriction feature

Prerequisites for administering Class of Restriction

You must complete the following actions before you can administer the Class of Restriction feature:

- None

Screens for administering Class of Restriction

Screen name	Purpose	Fields
<i>Class of Restriction Information</i>	List what CORs are administered on your system.	All
<i>Class of Restriction</i>	• Apply administration settings to all objects that share the same COR number. • Restrict the types of calls that a user can make and receive. • Identify what CORs can be service observed, or be a service observer.	All
<i>Optional Features</i>	Ensure that users can change their own COR without assistance from an administrator.	Change COR by FAC
<i>Feature Access Code (FAC)</i>	Assign a feature access code (FAC) so that the user can change their own COR without assistance from an administrator.	Change COR Access Code
<i>Feature-Related System Parameters</i>	Assign a password that is required before a user can change their own COR.	Password to Change COR by FAC

Displaying administered CORs

To list what CORs are administered on your system:

- 1 Type **list cor**. Press **Enter**.

The system displays the ***Class of Restriction Information*** screen ([Figure 132, Class of Restriction Information screen](#), on page 537).

Figure 132: Class of Restriction Information screen

list cor	CLASS OF RESTRICTION INFORMATION		Page 1 of 7
	COR	COR Description	
	0		
	1		
	2		
	3		
	4		
	5		
	6		
	7		
	8		
	9		
	10	Supervisor observer	
	11		
	12		
	13		
	14		

- 2 Press **Next** continually to see all the CORs.
- 3 Press **Cancel** when you finish.

Setting up a COR

To set up a COR:

- 1 Type **change cor *n***, where *n* is the number of a specific COR. Press **Enter**.
The system displays the *Class of Restriction* screen ([Figure 133, Class of Restriction screen, page 1](#), on page 538).

Figure 133: Class of Restriction screen, page 1

change cor 10	CLASS OF RESTRICTION		Page 1 of 4
COR Number: 10			
COR Description: Supervisor observer			
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		
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		

- 2 In the **COR Description** field, type a name for this COR. Assign a name for the COR that clearly reflects the purpose or members of the COR.
- 3 Complete all the applicable fields on page 1 of this screen. Specifically, you must:
 - Right-click the **Calling Party Restriction** field to see a list of options. Select an appropriate item from the list. For more information, see [Types of Restrictions](#) on page 532.
 - Right-click the **Called Party Restriction** field to see a list of options. Select an appropriate item from the list. For more information, see [Types of Restrictions](#) on page 532.
- 4 Press **Enter** to save your changes.
- 5 Press **Next** to see the next screen ([Figure 134, Class of Restriction screen, page 2](#), on page 539).

Figure 134: Class of Restriction screen, page 2

change cor 10	CLASS OF RESTRICTION	Page 2 of 4
MF Incoming Call Trace: n Brazil Collect Call Blocking? n Block Transfer Display? n Block Exchanged Conference/Transfer Display? n Remote Logout of Agent? n Station Lock COR: 9 Outgoing Trunk Disconnect Timer (minutes):		

- 6** Complete all the applicable fields on page 2 of this screen.
- 7** Press **Enter** to save your changes.
- 8** Press **Next** to see the next screen ([Figure 135, Class of Restriction screen, page 3](#), on page 539).
The numbers represent the 96 available CORs, from 0 to 95. All fields default to **y**.

Figure 135: Class of Restriction screen, page 3

change cor 10	CLASS OF RESTRICTION	Page 3 of 4																																																																																																
CALLING PERMISSION (Enter "y" to grant permission to call specified COR) <table style="width: 100%; border-collapse: collapse;"> <tr><td>0? y</td><td>12? y</td><td>24? y</td><td>36? y</td><td>48? y</td><td>60? y</td><td>72? y</td><td>84? y</td></tr> <tr><td>1? y</td><td>13? y</td><td>25? y</td><td>37? y</td><td>49? y</td><td>61? y</td><td>73? y</td><td>85? y</td></tr> <tr><td>2? y</td><td>14? y</td><td>26? y</td><td>38? y</td><td>50? y</td><td>62? y</td><td>74? y</td><td>86? y</td></tr> <tr><td>3? y</td><td>15? y</td><td>27? y</td><td>39? y</td><td>51? y</td><td>63? y</td><td>75? y</td><td>87? y</td></tr> <tr><td>4? y</td><td>16? y</td><td>28? y</td><td>40? y</td><td>52? y</td><td>64? y</td><td>76? y</td><td>88? y</td></tr> <tr><td>5? y</td><td>17? y</td><td>29? y</td><td>41? y</td><td>53? y</td><td>65? y</td><td>77? y</td><td>89? y</td></tr> <tr><td>6? y</td><td>18? y</td><td>30? y</td><td>42? y</td><td>54? y</td><td>66? y</td><td>78? y</td><td>90? y</td></tr> <tr><td>7? y</td><td>19? y</td><td>31? y</td><td>43? n</td><td>55? y</td><td>67? y</td><td>79? y</td><td>91? y</td></tr> <tr><td>8? y</td><td>20? y</td><td>32? y</td><td>44? y</td><td>56? y</td><td>68? y</td><td>80? y</td><td>92? y</td></tr> <tr><td>9? y</td><td>21? y</td><td>33? y</td><td>45? y</td><td>57? y</td><td>69? y</td><td>81? y</td><td>93? y</td></tr> <tr><td>10? y</td><td>22? y</td><td>34? y</td><td>46? y</td><td>58? y</td><td>70? y</td><td>82? y</td><td>94? y</td></tr> <tr><td>11? y</td><td>23? y</td><td>35? y</td><td>47? y</td><td>59? y</td><td>71? y</td><td>83? y</td><td>95? y</td></tr> </table>			0? y	12? y	24? y	36? y	48? y	60? y	72? y	84? y	1? y	13? y	25? y	37? y	49? y	61? y	73? y	85? y	2? y	14? y	26? y	38? y	50? y	62? y	74? y	86? y	3? y	15? y	27? y	39? y	51? y	63? y	75? y	87? y	4? y	16? y	28? y	40? y	52? y	64? y	76? y	88? y	5? y	17? y	29? y	41? y	53? y	65? y	77? y	89? y	6? y	18? y	30? y	42? y	54? y	66? y	78? y	90? y	7? y	19? y	31? y	43? n	55? y	67? y	79? y	91? y	8? y	20? y	32? y	44? y	56? y	68? y	80? y	92? y	9? y	21? y	33? y	45? y	57? y	69? y	81? y	93? y	10? y	22? y	34? y	46? y	58? y	70? y	82? y	94? y	11? y	23? y	35? y	47? y	59? y	71? y	83? y	95? y
0? y	12? y	24? y	36? y	48? y	60? y	72? y	84? y																																																																																											
1? y	13? y	25? y	37? y	49? y	61? y	73? y	85? y																																																																																											
2? y	14? y	26? y	38? y	50? y	62? y	74? y	86? y																																																																																											
3? y	15? y	27? y	39? y	51? y	63? y	75? y	87? y																																																																																											
4? y	16? y	28? y	40? y	52? y	64? y	76? y	88? y																																																																																											
5? y	17? y	29? y	41? y	53? y	65? y	77? y	89? y																																																																																											
6? y	18? y	30? y	42? y	54? y	66? y	78? y	90? y																																																																																											
7? y	19? y	31? y	43? n	55? y	67? y	79? y	91? y																																																																																											
8? y	20? y	32? y	44? y	56? y	68? y	80? y	92? y																																																																																											
9? y	21? y	33? y	45? y	57? y	69? y	81? y	93? y																																																																																											
10? y	22? y	34? y	46? y	58? y	70? y	82? y	94? y																																																																																											
11? y	23? y	35? y	47? y	59? y	71? y	83? y	95? y																																																																																											

- 9** To restrict a user who is assigned this COR from calling someone in another COR, change the Calling Permission of the COR number to **n**.
In this example, a user who is assigned a COR 10 cannot call a user who is assigned a COR 43.

- 10** Press **Enter** to save your changes.
- 11** Press **Next** to see the last screen ([Figure 136, Class of Restriction screen, page 4](#), on page 540).
The numbers represent the 96 available CORs, from 0 to 95. All fields default to **y**. Complete this screen only if you set the **Can Be A Service Observer** field on page 1 to **y**. If the **Can Be A Service Observer** field is set to **n**, you can skip this screen.
If a specified COR is set to **y**, but the **Can Be Service Observed** field on page 1 of that COR is set to **n**, that COR cannot be service observed.

Figure 136: Class of Restriction screen, page 4

change cor 10		CLASS OF RESTRICTION						Page 4 of 4
SERVICE OBSERVING PERMISSIONS (Enter "y" to grant permission to service observe the specified COR)								
0? y	12? y	24? y	36? y	48? y	60? y	72? y	84? y	
1? y	13? y	25? y	37? y	49? y	61? y	73? y	85? y	
2? y	14? y	26? y	38? y	50? y	62? y	74? y	86? y	
3? y	15? y	27? y	39? y	51? y	63? y	75? y	87? y	
4? y	16? y	28? y	40? y	52? y	64? y	76? y	88? y	
5? y	17? y	29? y	41? y	53? y	65? y	77? y	89? y	
6? y	18? y	30? y	42? y	54? y	66? y	78? y	90? y	
7? y	19? n	31? y	43? y	55? y	67? y	79? y	91? y	
8? y	20? y	32? y	44? y	56? y	68? y	80? y	92? y	
9? y	21? y	33? y	45? y	57? y	69? y	81? y	93? y	
10? y	22? y	34? y	46? y	58? y	70? y	82? y	94? y	
11? y	23? y	35? y	47? y	59? y	71? y	83? y	95? y	

- 12** To indicate what COR cannot be service observed, change the value of the COR number to **n**.
In this example, a user who is assigned a COR 10 cannot service observe a user who is assigned a COR 19.
- 13** Press **Enter** to save your changes.

Allowing users to change their own COR

You can allow users to change their own COR from the telephone through a feature access code (FAC). To restrict this feature, you can also require that users enter a password before the users can change their own COR.

To allow users to change their own Class of Restriction, you must complete the following procedures:

- [Assigning a Feature Access Code \(FAC\)](#)
- [Assigning a password](#)

Prerequisites

You must complete the following actions before you can allow users to change their own COR:

- On the **Optional Features** screen, ensure that the Change COR by FAC field is set to **y**. If this field is not set to **y**, users cannot change their own COR. Contact your Avaya representative for assistance.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

For a complete description of the many **Optional Features** screens, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Assigning a Feature Access Code (FAC)

To assign a FAC:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the **Feature Access Code (FAC)** screen ([Figure 137, Feature Access Code \(FAC\) screen](#), on page 541).

Figure 137: Feature Access Code (FAC) screen

change feature-access-codes	Page 1 of 8
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: ____ Attendant Access Code: ____ Auto Alternate Routing (AAR) Access Code: ____ Auto Route Selection (ARS) - Access Code 1: ____ Automatic Callback Activation: ____ Call Forwarding Activation Busy/DA: ____ All: ____ 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: *55 Change Coverage Access Code: ____ Contact Closure Open Code: ____ Contact Closure Pulse Code: ____ Access Code 2: ____ Deactivation: ____ Deactivation: ____ Close Code: ____	

- 2 In the Change COR Access Code field, type a FAC that conforms to your dial plan.
- 3 Press **Enter** to save your changes.
- 4 Ensure that you notify all users of the assigned FAC.

Assigning a password

To assign a password:

- 1 Type **change system-parameters features**. Press **Enter**.
The system displays the *Feature-Related System Parameters* screen.
- 2 Click **Next** until you see the Password to Change COR by FAC field ([Figure 138, Feature-Related System Parameters screen](#), on page 542).

Figure 138: Feature-Related System Parameters screen

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	

-
- 3 In the Password to Change COR by FAC field, type a password.
This field determines if Communication Manager requires the user to enter a password when the user tries to change the COR, and what that password is.
 - 4 Press **Enter** to save your changes.
 - 5 Ensure that you notify all users of the assigned password.

End-user procedures for Class of Restriction

End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for the Class of Restriction feature:

To temporarily change your COR from your telephone through a feature access code (FAC):

- 1 Dial the access code FAC to change your COR.
- 2 Dial the password to change your COR.

Reports for Class of Restriction

The following reports provide information about the Class of Restriction feature:

- None

Considerations for Class of Restriction

This section provides information about how the Class of Restriction feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Class of Restriction under all conditions.

The following considerations apply to the Class of Restriction feature:

- None

Interactions for Class of Restriction

This section provides information about how the Class of Restriction feature interacts with other features in your system. Use this information to ensure that you receive the maximum benefits of the Class of Restriction in any feature configuration.

- Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS)
Termination and miscellaneous restrictions do not apply to AAR or ARS calls. AAR or ARS access to a trunk group overrides miscellaneous trunk restrictions.
- AAR or ARS Partitioning
Use a COR to assign partition group numbers.
- Abbreviated Dialing
The system does not check direct calls that are made from group lists. These lists are inaccessible to a user whose telephone is fully restricted. However, once a user makes a call from the Abbreviated Dialing group list, the system checks all subsequent transfer and conference attempts.
- 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. The system does not check COR assignments.
- Bridged Call Appearance
A COR that is assigned to a telephone also applies to calls that originate from a bridged call appearance of that telephone.
- Call Coverage
Users who are restricted from calls can still receive calls that Call Coverage directs to these users. When a call goes to coverage, the system uses the restrictions of the called party, not of the covering party. When a call is redirected to coverage, the system does not check the COR of the covering party. If the COR of the covering party is fully restricted, the system cannot complete the call.
- Call Forwarding All Calls
If a COR restricts a call between the forwarding extension and the forwarded-to extensions, Call Forwarding is denied. The system always checks when Call Forwarding is activated, but not when the system forwards a call.

- **Call Vectoring**

When the system directs a call to a vector directory number (VDN), the system compares the COR of the caller and the VDN. This comparison determines if the caller can access the associated call vector.
- **Centralized Attendant Service (CAS)**

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 be completed to a fully restricted telephone.
- **Class of Restriction Display to the Attendant**

The attendant can display the COR for each telephone.
- **Class of Service (COS)**

In some cases, the Class of Service (COS) of a user can override a COR. For more information, see the **Trk-to-Trk Restriction Override** field in the Class of Service (COS) feature description.
- **Controlled Restriction**

Restrictions that you assign through Controlled Restriction override any COR restrictions.
- **Distributed Communications System (DCS)**

Fully restricted service allows all DCS calls. DCS can allow a public network call to be completed to a fully restricted telephone.
- **Emergency Access to Attendant**

A COR does not restrict Emergency Access to Attendant calls.
- **Fully Restricted Service**

Do not assign fully restricted service to a telephone with the following features or conditions:

 - Abbreviated Dialing
 - Bridged Call Appearance
 - Attendant telephones
 - Night Service telephones
 - Telephones that are Call Coverage or Send All Calls points
 - Telephones that are Call Forward destinations
 - Telephones that are Call Pickup points
- **Hunt Groups**

The system checks the COR that is assigned to a hunt group on calls that are redirected by the DDC or UCD of the hunt group. If the COR of the hunt group does not have fully restricted service, extensions in the hunt group can receive calls from the public network.
- **Night Service and Night Station Trunk Answer From Any Station**

Both the Night Service feature and the Night Station Trunk Answer From Any Station feature override:

 - Inward restrictions
 - Manual terminating line restrictions
 - Public restrictions

- **Personal Central Office Line (PCOL)**
Do not assign fully restricted service to users who have a Personal Central Office Line. If you do, you are paying for a CO line that no one can use.
- **Power Failure Transfer**
All authorization features are bypassed when the system is in emergency transfer mode.
- **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. The system does not check COR assignments.
- **Remote Access**
If the user enters a barrier code during connection to remote access, the system uses the COR that is associated with that code for authorization checks. If remote access does not require a barrier code, then the system uses the COR of the default barrier code. Remote access can require an authorization code instead of, or in addition to, the barrier code. If the system requires an authorization code, the COR of the authorization code overrides the COR of the barrier code.
- **Tie trunk access**
The system can complete incoming dial-repeating tie trunk calls directly to an inward-restricted telephone or a public-restricted telephone. An attendant cannot extend incoming dial-repeating tie trunk calls to an inward-restricted telephone.
- **Transfer**
When you administer a three-way conference, a user cannot transfer incoming trunk calls to an inward-restricted telephone. Transferred calls are subject to 3-way COR Checking restrictions.
A user can transfer incoming trunk calls from an unrestricted telephone to an inward-restricted telephone or a public-restricted telephone. However, you must override the three-way conference COR.

Class of Service

Use the Class of Service (COS) feature to allow or deny user access to some system features.

Detailed description of Class of Service

This section provides a detailed description of the Class of Service (COS) feature.

Use the COS feature to allow or deny user access to some system features, such as:

- Automatic Callback
- Call Forwarding
- Data Privacy
- Trunk-to-Trunk Transfer Override
- QSIG Call Offer Originations
- Contact Closure Activation
- Console Permission

Use the Class of Restriction (COR) feature, instead of COS, to define the restrictions that apply when a user places or receives a call. For more information, see the “Class of Restriction” feature.

COS does not apply to trunk groups, except for the Remote Access feature. For more information, see the “Remote Access” feature.

Hardware requirements for Class of Service

The Class of Service feature requires the following hardware:

- None

Administering Class of Service

The following steps are part of the administration process for the Class of Service feature (COS):

- [Defining COS for your system](#)
- [Assigning a COS](#)

This section describes:

- Any prerequisites for administering the Class of Service feature
- The screens that you use to administer the Class of Service feature
- Complete administration procedures for the Class of Service feature

Prerequisites for administering Class of Service

You must complete the following actions before you can administer the Class of Service feature:

- None

Screens for administering Class of Service

Screen name	Purpose	Fields
<i>Attendant Console</i>	Assign Class of Service (COS) for the attendant.	COS
<i>Class of Service</i>	Define COS for your system.	All
<i>Console-Parameters</i>	Assign COS for all the attendant consoles.	COS
<i>Data Modules</i>	Assign COS for a data module.	COS
<i>Remote Access</i>	Assign the COS associated with the barrier code of the Remote Access extension.	COS
<i>Station</i>	Assign COS for a user.	COS

Defining COS for your system

To define COS for your system:

- 1 Type **change cos.** Press **Enter**.

The system displays the **Feature Access Codes** screen ([Figure 139, Class of Service screen](#), on page 549).

Figure 139: Class of Service screen

change cos	CLASS OF SERVICE															Page 1 of 1
	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 (PSA)	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

- 2 Perform one of the following actions for any, or all, of the COSs, which are numbered zero through 15:
 - If you want to activate the feature for the COS, type **y**.
 - If you do not want to activate the feature for the COS, type **n**.
- 3 Press **Enter** to save your changes.

Descriptions of the COS features

The system displays some features only if the associated feature is set to **y** on the **Feature-Related System Parameters** screen. The information on the **Class of Service** screen that you display can differ from the information shown in the [Figure 139, Class of Service screen](#), on page 549.

- **Automatic Callback**
Allows a user to request Automatic Callback. For more information, see the “Automatic Callback” feature.
- **Automatic Exclusion**
Allows a user to automatically activate Privacy Exclusion when the user goes off hook at a telephone that has an assigned Exclusion button.
If you set this field to **n**, the user can use manual exclusion when the user presses the Exclusion button, either before the user dials a call or during a call.
The system displays this field when the Automatic Exclusion by COS field on the **Feature-Related System Parameters** screen is set to **y**.
- **Call Forwarding All Calls**
Allows a user to forward all calls to any extension. For more information, see the “Call Forwarding” feature.

- **Call Forwarding Busy/DA**

Allows the system to forward calls when the user is active on a call, or does not answer a call. For more information, see the “Call Forwarding” feature.

- **Client Room**

Allows users to access the 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 a COS for Client Room only when you have Hospitality Services and a Property Management System (PMS) interface. See the *GuestWorks® and DEFINITY® Systems Technician Handbook for Hospitality Installations* for more information.

- **Console Permissions**

Console Permissions allows a user who has a multiappearance telephone, 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

- **Data Privacy**

Allows a user to enter a feature access code (FAC) to protect a data call from interruption by any of the system override or ringing features. For more information, see the “Privacy” feature.

- **Extended Forwarding All**

Allows a user to use Call Forwarding All Calls from an off-site telephone.

You can change a COS to **y**, only if the **Extended Cvg/Fwd Admin** field on the **System Parameters Customer-Options** screen is set to **y**. For more information, see the “Extended User Administration of Redirected Calls” feature.

- **Extended Forwarding B/DA**

Allows a user to activate Call Forwarding from an off-site telephone.

You can change a COS to **y** only if the **Extended Cvg/Fwd Admin** field on the **System Parameters Customer-Options** screen is set to **y**. For more information, see the “Extended User Administration of Redirected Calls” feature.

- **Off-Hook Alert**

You can change a COS to **y**, only if either the **Hospitality (Basic)** field or the **Emergency Access to Attendant** field on the **System-Parameters Customer-Options** screen is set to **y**. For more information, see the “Emergency Access to Attendant” feature.

- **Personal Station Access (PSA)**
Allows a user to use an FAC to associate a telephone to the extension that is assigned to the user.
You must set this field to **n** for virtual telephones.
You can change this field to **y**, only if the **Personal Station Access (PSA)** field on the **Optional Features** screen is set to **y**. For more information, see the “Personal Station Access” feature.
- **Priority Calling**
Allows a user to dial a FAC to originate a priority call. For more information, see the “Priority Calling” feature.
- **QSIG Call Offer Originations**
Allows a user to invoke QSIG Call Offer services. For more information, see *Administration for Network Connectivity for Avaya™ Communication Manager*.
- **Restrict Call Fwd-Off Net**
If you set this field to **y**, a user cannot forward calls to the public network.
For security reasons, type **y** in the **Restrict Call Fwd-Off Net** field for all COS, except those that you use for special circumstances. For more information, see the “Call Forwarding” feature.
- **Trk-to-Trk Restriction Override**
Allows a user to override any system COR-to-COR calling party restrictions that would otherwise prohibit the trunk-to-trunk transfer operation for a user with this COS. For more information, see the “Transfer” feature.



SECURITY ALERT:

You increase the risk of toll fraud, if you allow users to perform trunk-to-trunk transfers.

Assigning a COS

To assign a COS:

- 1 Change the COS field on any of the following screens:
 - *Attendant Console*
 - *Console-Parameters*
 - *Data Modules*
 - *Remote Access*
 - *Station*
- 2 Press **Enter** to save your changes.

Reports for Class of Service

The following reports provide information about the Class of Service feature:

- None

Considerations for Class of Service

This section provides information about how the Class of Service feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Class of Service under all conditions. The following considerations apply to Class of Service:

- Hunt Groups

Many Hunt Groups have a COS of 1. Ensure that you do not cause unintended restrictions for Hunt Groups when you administer COS 1.

Interactions for Class of Service

This section provides information about how the Class of Service feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Class of Service in any feature configuration.

- Class of Service (COS) controls use of the following features and capabilities:
 - Automatic Callback
 - Automatic Exclusion
 - Call Forwarding
 - Call Forward Busy/Don't Answer
 - Client room
 - Console permission
 - Data Privacy
 - Extended Forwarding All
 - Extended Forwarding Busy/Don't Answer
 - Off-hook alert
 - Personal Station Access
 - Priority Calling
 - QSIG Call Offer Originations
 - Restrict Call Forwarding Off-Net
 - Trunk-to-Trunk Transfer Restriction Override

Co-resident DEFINITY LAN Gateway

Use the Co-resident DEFINITY LAN Gateway (DLG) feature to provide the functionality of the Adjunct/Switch Application Interface (ASAI), with the use of an Ethernet transport instead of the traditional Basic Rate Interface (BRI) transport.

Detailed description of Co-resident DEFINITY LAN Gateway

This section provides a detailed description of the Co-resident DEFINITY LAN Gateway (DLG) feature.

Co-resident DEFINITY LAN Gateway is a software application that is part of Avaya MultiVantage™ Communications Applications on the server that is running Avaya Communication Manager. In the S8300 Media Server with a G700 Media Gateway, Ethernet connectivity is provided through the processor.

The DEFINITY LAN Gateway software enables communications between TCP/IP clients and Avaya Communication Manager call processing. The DLG application software:

- Routes internetwork messages from one protocol to another, for example, ISDN to TCP/IP
- Bridges all Adjunct/Switch Application Interface (ASAI) message traffic by way of a TCP/IP tunnel protocol.

In previous configurations, a DEFINITY LAN gateway (DLG) was connected externally on a separate TN801 MAPD circuit pack. The DLG application is now packaged internally, where the application co-resides with the Communication Manager software. The internally packaged DLG is referred to as the Co-resident DLG. Co-resident DLG is available only with the S8300 Media Server. Co-resident DLG provides the functionality of the Adjunct/Switch Application Interface (ASAI), with the use of an Ethernet transport instead of a Basic Rate Interface (BRI) transport.

How Co-resident DLG works

- DLG listens for client connections over a well-known TCP port (5678) at a specified IP address.
- The client connects to TCP port 5678 at the IP address to receive the services of the DLG.
- The client exchanges TCP Tunnel Protocol messages with the DLG to request a connection to a specific Computer Telephony Integration (CTI) link.
- DLG authenticates the client based on the administration of the client and then establishes or refuses the connection.
- Once the connection is established, the actual ASAI layer 3 messages are passed transparently through the DLG. Each TCP connection to the DLG has a one-to-one correspondence with a CTI link.

Hardware requirements for Co-resident DEFINITY LAN Gateway

The Co-resident DEFINITY LAN Gateway feature requires the following hardware:

CallVisor ASAI CTI link

- An Ethernet interface for connectivity to adjuncts. The platform configuration determines the Ethernet interface.
 - The following platforms use the TN801B MAPD-based DEFINITY LAN Gateway (DLG) as the Ethernet interface:
 - DEFINITY CSI
 - DEFINITY SI
 - DEFINITY R
 - For S8300 Media Server configurations with a G700 that have an Internal Communications Controller, or ICC, Co-resident DLG relies on the S8300 Media Server as the Ethernet interface.
- Packet Controller circuit pack (for internal communications)
- Packet Maintenance circuit pack

Co-resident DLG

- Supported platforms:
 - S8300 Media Server with a G700
- Unsupported platforms:
 - DEFINITY CSI
 - DEFINITY SI
 - DEFINITY R
 - S8700 IP-Connect
 - S8700 Multi-Connect

Administering Co-resident DEFINITY LAN Gateway

This section describes:

- Any prerequisites for administering the Co-resident DEFINITY LAN Gateway feature
- The screens that you use to administer the Co-resident DEFINITY LAN Gateway feature

Prerequisites for administering Co-resident DEFINITY LAN Gateway

You must complete the following actions before you can administer the Co-resident DEFINITY LAN Gateway feature:

TN799 C-LAN circuit pack as the Ethernet Interface

- View the **Optional Features** screen, and ensure that the ASAI Link Core Capabilities field, the Computer Telephony Adjunct Links field, or both fields are set to **y**, and that the Co-Res DEFINITY LAN Gateway field is set to **y**. If these fields are set to **n**, your system is not enabled for the Co-resident DEFINITY LAN Gateway feature. Contact your Avaya representative before you continue with this procedure.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

- View the **IP Node Names** screen, and ensure that a node name is associated with the C-LAN. If a node name is not associated with the C-LAN, add a node name.

To view the **IP Node Names** screen, type **display node-names ip**. To add a node name, type **change node-names ip**. Press **Enter**.

- View the **IP Interfaces** screen, and ensure that a C-LAN is administered, and the Ethernet port for the C-LAN is enabled. If a C-LAN is not listed, add a C-LAN.

To view the **IP Interfaces** screen, type **display ip-interfaces**. To add a C-LAN, type **add ip-interface next**. Press **Enter**.

- View the **Data Modules** screen, and ensure that the Ethernet port on the C-LAN is administered. If an Ethernet port on the C-LAN is administered, the **Service Type** field is set to **Ethernet**, and the **Port** field is set to port **17** on the TN799. If the Ethernet port is not listed, add an Ethernet port.

To view the **Data Modules** screen, type **list data-modules**. To add an Ethernet port, type **add data-module next**. Press **Enter**.

TN2314 circuit pack as the Ethernet Interface

- View the **Optional Features** screen, and ensure that the ASAI Link Core Capabilities field, the Computer Telephony Adjunct Links field, or both fields, are set to **y**. Click **Next** until the Processor Ethernet field appears. Ensure that the Processor Ethernet field is set to **y**. If these fields are set to **n**, your system is not enabled for the Co-resident DEFINITY LAN Gateway feature. Contact your Avaya representative before you continue with this procedure.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

- View the **IP Interfaces** screen, and ensure that the processor (PROCR) is administered and the Ethernet port for the processor is enabled. If the processor is administered, and the Ethernet port is enabled, **PROCR** appears in the **Type** field and the previously administered node name appears in the **Node Name** field. If the PROCR is not listed, add a processor.

To view the **IP Interfaces** screen, type **display ip-interfaces**. To add a processor, type **add ip-interface next**. Press **Enter**.

Screens for administering Co-resident DEFINITY LAN Gateway

Screen name	Purpose	Fields
<i>Optional Features</i>	Ensure that the proper license fields are set to y .	• ASAI Link Core Capabilities (if required) • Computer Telephony Adjunct Links (if required) • Co-Res DEFINITY LAN Gateway • Processor Ethernet
<i>IP Node Names</i>	Ensure that a node name is associated with the C-LAN.	• Name
<i>IP Interfaces</i>	Ensure that the C-LAN is administered, and the Ethernet port for the C-LAN is enabled. If the C-LAN is administered and the Ethernet port is enabled, the Type field is set to C-LAN , and the previously administered node name appears in the Node Name / Local Node field.	• Type • Node Name/Local Node
<i>Data Modules</i>	Ensure that the Ethernet port on the C-LAN is administered. If the Ethernet port on the C-LAN is administered, the Service Type field is set to Ethernet , and the Port field is set to port 17 on the TN799.	• Service Type • Port
<i>CTI Link</i>	Add a CTI link.	All
<i>IP Services</i>	Set the IP service type to DLG .	Service Type
<i>Administer DLG</i>	When the Service Type field is set to DLG , this page is added to the <i>IP Services</i> screen.	All

NOTE:

For more information on DLG administration for the S8300 Media Server, see the *Avaya™ CallVisor® ASAI Technical Reference*.

Reports for Co-resident DEFINITY LAN Gateway

The following reports provide information about the Co-resident DEFINITY LAN Gateway feature:

- None

Considerations for Co-resident DEFINITY LAN Gateway

This section provides information about how the Co-resident DEFINITY LAN Gateway feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Co-resident DEFINITY LAN Gateway under all conditions. The following considerations apply to Co-resident DEFINITY LAN Gateway:

- None

Interactions for Co-resident DEFINITY LAN Gateway

This section provides information about how the Co-resident DEFINITY LAN Gateway feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Co-resident DEFINITY LAN Gateway in any feature configuration.

- None

Conference

Use the Conference feature, with the associated Conf button, to create a conference without the assistance of an attendant.

Conference supports the following capabilities:

- **Conference/Transfer Toggle/Swap**
Talk back and forth between two users before the user connects all the participants to the conference call.
- **No Dial Tone Conferencing**
Eliminate the dial tone that a user usually hears while other participants are added to a conference call.
- **No Hold Conference**
Add another user to a conference call, and not disrupt the call that is currently active.
- **Select Line Appearance Conferencing**
Use the line appearance button instead of pressing the Conf button a second time to complete the conference call.
- **Selective Conference Party Display, Drop, and Mute**
Identify the participants on a call.

The Meet-me Conference feature allows users to set up a dial-in conference of up to six parties. The Meet-me Conference feature uses Call Vectoring to process the setup of the conference call. For more information, see the “Meet Me Conference” feature.

Detailed description of Conference

This section provides a detailed description of the Conference feature.

Use the Conference feature, with the associated Conf button, to create a conference without the assistance of an attendant.

A user with a multiple appearance telephone with a Conf button, can create a conference with as many as six participants.

A user with a single-line telephone can create a conference with as many as three participants. Each of these three participants can then add another participant. Thus, a user who has a single-line telephone can create a conference call with as many as six participants.

Conference and DCP, hybrid, IP, wireless, and ISDN-BRI telephones

A user with a Digital Communications Protocol (DCP), hybrid, IP, wireless, or ISDN-BRI telephones can use the Conference feature for a call on hold when:

- Only one call is on hold
- No call appearances are active
- An available call appearance exists for the conference call

If more than one call is on hold, the user must make a call active to start a conference. If the user presses the Conf button when two or more calls are on hold, the system ignores the conference request from the user.

If the user has an active call, and also has calls on hold, the system includes the active call in the conference when the user presses the Conf button.

Conference/Transfer Toggle/Swap

A user who sets up a conference call can use the Conference/Transfer Toggle/Swap capability to talk back and forth between two users before the user connects all the participants to the conference call. The display also toggles between the two parties. The Conference/Transfer Toggle/Swap capability is unavailable on attendant consoles.

The user uses an administered feature button, toggle-swap, for the Conference/Transfer Toggle/Swap capability

No Dial Tone Conferencing

A user uses the No Dial Tone Conferencing capability to add existing calls to a conference call. The user does not need to select a line appearance if there is someone on hold, or if the system is alerting a line appearance. This capability eliminates the dial tone that the user usually hears while the user adds the participants to the conference call.

For example, a user placed a call on hold, and is now talking to another party. When the user presses the conf button, and then presses the button of the call on hold, the party on hold joins the conference.

No Hold Conference

When a user who is active on a call creates a conference call, the user can use the No Hold Conference capability to add another participant to a conference call, while the user continues a conversation with the participant on the call that is currently active. When the user calls the new participant, the new participant automatically joins the conference when the new participant answers the call.

For example, a user presses the administered no-hold-conf feature button and then dials an extension. The party that answers the call automatically joins the conference.

If the called user does not answer the call within the time that you specify in the **No Hold Conference Timeout** field on the **Feature-Related System Parameters** screen, the system deactivates the No Hold Conference capability for the call.

Users with multiline digital telephones can use the No Hold Conference capability.

Select Line Appearance Conferencing

If a user is at a call on line B, and another line is on hold or an incoming call alerts at line A, the user can press the Conf button to bridge the calls together. If the user uses the select line appearance capability, the user can press a line appearance button to complete a conference instead of pressing the Conf button a second time.

Selective Conference Party Display, Drop, and Mute

A user who has a digital telephone with a display, or an attendant with an attendant console, can use the Selective Conference Party Display, Drop, and Mute capability to identify the participants on a call.

The conference prompts that the system displays are based on the user Class of Restriction (COR). The display prompts are based on the user COR, independent of the select line appearance conferencing and the No Dial Tone Conferencing capability. The display messages vary depending on the activation of the various Conference capabilities. The user COR controls the display of any additional information.

A user presses the administered conf-dsp feature button to scroll through the telephone numbers and names of the participants on the call. The telephone numbers of the participants are always available, although the names of the participants are sometimes unavailable. When the system displays the telephone number of a participant, the user can either press the administered fe-mute feature button to place the participant on mute, or the user can press the Drop button to drop the participant from the call.

The ability to place a call participant on mute is useful when that participant has a noisy trunk line because of a cell phone, music-on-hold, or background noise. The Selective Conference Party Mute capability applies only to trunk lines, and a user can mute only one trunk line on a conference call.

Hardware requirements for Conference

The Conference feature requires the following hardware:

- None

Administering Conference

The following steps are part of the administration process for the Conference feature:

- [Assigning the toggle-mute feature button](#)
- [Assigning the conf-dsp, fe-mute, and no-hold-conf feature buttons](#)

This section describes:

- Any prerequisites for administering the Conference feature
- The screens that you use to administer the Conference feature
- Complete administration procedures for the Conference feature

Prerequisites for administering the Conference feature

You must complete the following actions before you can administer the Conference feature:

- Administer the Feature-Related System Parameters screen to support the Conference feature on your system.

To administer the Feature-Related System Parameters screen to support the Conference feature on your system:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the *Feature-Related System Parameters* screen ([Figure 140, Feature-Related System Parameters screen](#), on page 564) and ([Figure 141, Feature-Related System Parameters screen](#), on page 565).

Figure 140: Feature-Related System Parameters screen

change system-parameters features	Page 5 of 13
FEATURE-RELATED SYSTEM PARAMETERS	
Public Network Trunks on Conference Call: 5	Auto Start? n
Conference Parties with Public Network Trunks: 6	Auto Hold? y
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? y
Unanswered DID Call Timer (seconds):	Intrusion Tone? n
Line Intercept Tone Timer (seconds): 30	Special Dial Tone? n
Long Hold Recall Timer (seconds): 0	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: tone	
Allow AAR/ARS Access from DID/DIOD? n	
Allow ANI Restriction on AAR/ARS? n	
Use Trunk COR for Outgoing Trunk Disconnect? n	
7405ND Numeric Terminal Display? y	7434ND? n
DISTINCTIVE AUDIBLE ALERTING	
Internal: 1 External: 2 Priority: 3	
Attendant Originated Calls: external	

Figure 141: Feature-Related System Parameters screen

change system-parameters features	Page 6 of 13
FEATURE-RELATED SYSTEM PARAMETERS	
CONFERENCE/TRANSFER	
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? n	
AUDIX ONE-STEP RECORDING	
	Recording Delay Timer (msec): 500
Apply Ready Indication Tone To Which Parties In The Call? all	
Interval For Applying Periodic Alerting Tone (seconds): 15	

- 2 Page through the screens until you see the **Public Network Trunks on Conference Call** field.
- 3 In the **Public Network Trunks on Conference Call** field, type the maximum number of participants with public network trunks that you want to participate in a conference call. The valid entries for this field are **0** through **5**.
You can type **0**, if you do not want any public network trunks to participant in a conference call. If you type **0** in this field, the **Conference Parties with Public Network Trunks** field does not appear on this **Feature-Related System Parameters** screen.
- 4 In the **Conference Parties without Public Network Trunks** field, type the maximum number of participants without public network trunks that you want to participate in a conference call. The valid entries for this field are **3** through **5**.
- 5 In the **Conference Tone** field, perform one of the following actions:
 - Type **y** if you want the participants in a conference call to hear the conference tone if there are three or more participants on a conference call.
 - Type **n** if you do not want conference participants to hear the conference tone if there are three or more participants on a conference call.
- 6 Page through the screens until you see the **No Dial Tone Conferencing** field.
- 7 In the **No Dial Tone Conferencing** field, take one of the following actions:
 - Type **n**, if you want a user who is on hold to hear dial tone while the conference owner adds another conference participant.
 - Type **y**, if you do not want a user who is on hold to hear dial tone while the conference owner adds another conference participant.
- 8 In the **Select Line Appearance Conferencing** field, take one of the following actions:
 - Type **y** to activate select line appearance conferencing.
 - Type **n** to deactivate select line appearance conferencing.

- 9** In the **Abort Conference on Hang-up** field, perform one of the following actions:
 - Type **y** if you want the user to stop a conference call if the user hangs up the telephone before the conference call is complete.
 - Type **n** if you do not want the user to stop a conference call when the user hangs up the telephone before the conference call is complete.
- 10** In the **No Hold Conference Timeout** field, type the number of seconds that you want the system to wait while a user uses the No Hold Conference capability to add a participant to a conference call. Note that you must set the **Answer Supervision** field on the **Trunk Group** screen to fewer seconds than the seconds in the **No Hold Conference Timeout** field.
- 11** Press **Enter** to save your changes.

Screens for administering Conference

Screen name	Purpose	Fields
<i>Attendant Console</i>	Assign feature buttons to an attendant console.	Feature Button Assignments: • conf-dsp • fe-mute

Screen name	Purpose	Fields
Feature-Related System Parameters	Allow users of DCP, hybrid, IP, wireless, or ISDN-BRI telephones to “abort the conference operation when they hang up.”	Abort Conference Upon Hang-Up
	Specify that the system generate a conference tone.	Conference Tone
	Specify the maximum number of participants in a conference call when any of the participants uses a public network trunk.	Conference Parties With Public Network Trunks
	Specify the maximum number of participants on a conference call when none of the participants uses a public network trunk.	Conference Parties Without Public Network Trunks
	Specify the number of seconds before the system deactivates the No Hold Conference capability for a call.	No Hold Conference Timeout
	Specify that a user who is on hold hears dial tone while the conference owner adds another conference participant.	No Dial Tone Conferencing
	Specify that the user can use the line appearance rather than the Conf button to include a call in a conference.	Select Line Appearance Conferencing
Station	Assign feature buttons to a user telephone, or an attendant console	Button Assignments: • conf-dsp • fe-mute • no-hold-conf • toggle-swap

Assigning the toggle-mute feature button

To assign the toggle-mute feature button to a user telephone:

- 1 Type **change station *n***, where ***n*** is the telephone number of the extension to which you want to assign the Conference/Transfer Toggle/Swap capability. Press **Enter**.

The system displays the **Station** screen for the extension that you requested ([Figure 144, Station screen](#), on page 570).

Figure 142: Station screen

change station 20001	STATION	Page 3 of 4
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: toggle-swap	
2: call-appr	6:	
3: call-appr	7:	
4:	8:	

- 2 Page through the screens until you see the **BUTTON ASSIGNMENTS** area.
- 3 In the **BUTTON ASSIGNMENTS** area, type **toggle-swap** next to the button that you want the user to use for the Conference/Transfer Toggle/Swap capability.
- 4 Press **Enter** to save your change.

Assigning the conf-dsp, fe-mute, and no-hold-conf feature buttons

Prerequisites

You must complete the following actions before you can assign the conf-dsp, fe-mute, and no-hold-conf feature buttons:

- On the **Optional Features** screen, verify that the **Enhanced Conferencing?** field is set to **y**, if you want to use the Selective Conference Party Display, Drop, and Mute capability and the No Hold Conference capability on your system. To view this screen, type **display system-parameters customer-options**. Press **Enter**. If the **Enhanced Conferencing?** field is set to **n**, your system is not enabled for the Selective Conference Party Display, Drop, and Mute capability. Contact your Avaya representative for assistance before you continue with this procedure.

For a complete description of the **Optional Features** screen, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

- Assign a COR for the Selective Conference Party Display, Drop, and Mute capability.

To Assign a COR for the Selective Conference Party Display, Drop, and Mute capability:

- 1 Type **change COR *n***, where *n* is the number of the COR to which you want to assign the Selective Conference Party Display, Drop, and Mute capability. Press **Enter**.

The system displays the **Class of Restriction** screen ([Figure 143, Class of Restriction screen](#), on page 569).

Figure 143: Class of Restriction screen

change cor 10	CLASS OF RESTRICTION	Page 2 of 4
	MF Incoming Call Trace? n Brazil Collect Call Blocking? n Block Transfer Display? n Block Enhanced Conference/Transfer Displays? y Remote Logout of Agent? n	
	Station Lock COR: 10 Outgoing Trunk Disconnect Timer (minutes):	

- 2 In the Block Enhanced Conference/Transfer Displays? field, perform one of the following actions:
 - Type **y** if you want the system to block all the enhanced conference/transfer display messages except “Transfer Completed” for a user or an attendant.
 - Type **n** if you do not want the system to block all the enhanced conference/transfer display messages except “Transfer Completed” for a user or an attendant.

To assign the conf-dsp, fe-mute, and no-hold-conf feature buttons, you must complete the following procedures:

- Assign the conf-dsp, fe-mute, and no-hold-conf feature buttons to a user.
- Assign the conf-dsp and fe-mute feature buttons to an attendant.

Assigning the conf-dsp, fe-mute, and no-hold-conf feature buttons to a user

To assign the conf-dsp, fe-mute, no-hold-conf feature buttons to a user telephone:

- 1 Type **change station *n***, where *n* is the telephone number of the extension to which you want to assign the following capabilities:
 - Conference Display
 - Selective Conference Party Display, Drop, and Mute
 - No-Hold Conference

Press **Enter**.

The system displays the **Station** screen for the extension that you requested ([Figure 144, Station screen](#), on page 570).

Figure 144: Station screen

change station 20001		Page 3 of 4
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: toggle-swap	
2: call-appr	6: conf-dsp	
3: call-appr	7: fe-mute	
4:	8: no-hold-conf	

- 2 Page through the screens until you see the **BUTTON ASSIGNMENTS** area.
- 3 In the **BUTTON ASSIGNMENTS** area, perform the following actions:
 - Type **conf-dsp** next to the button that you want the user to use for the Conference Display capability.
 - Type **fe-mute** next to the button that you want the user to use for the Selective Conference Party Display, Drop, and Mute capability.
 - Type **no-hold-conf** next to the button that you want the user to use for the No-Hold Conference capability.
- 4 Press **Enter** to save your changes.

Assigning the conf-dsp and fe-mute feature buttons to an attendant

To assign the conf-dsp and fe-mute feature buttons to an attendant console:

- 1 Type **change attendant *n***, where *n* is the number of the attendant console to which you want to assign the Selective Conference Party Display, Drop, and Mute capability. Press **Enter**.
The system displays the **Attendant Console** screen for the attendant that you requested ([Figure 145, Attendant Console screen](#), on page 571).

Figure 145: Attendant Console screen

change attendant 1	ATTENDANT CONSOLE	Page 3 of 4
FEATURE BUTTON ASSIGNMENTS		
1: split	13: conf-dsp	
2: dn-dst	14: fe-mute	
3: split-swap	15:	
4:	16:	
5:	17:	
6: hold	18:	
7: serial-cal	19: forced-rel	
8: verify	20:	
9: intrusion	21:	
10:	22:	
11:	23: night-serv	
12:	24: pos-busy	

- 2 Page through the screens until you see the FEATURE BUTTON ASSIGNMENTS area.
- 3 In the FEATURE BUTTON ASSIGNMENTS area, perform the following actions:
 - Type **conf-dsp** next to the button that you want the attendant to use for the Conference Display capability
 - Type **fe-mute** next to the button that you want the attendant to use for the Selective Conference Party Display, Drop, and Mute capability.
- 4 Press **Enter** to save your changes.

End-user procedures for Conference

End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Conference:

- [Displaying the participants on a conference call](#)

Displaying the participants on a conference call

Users of a digital telephone and attendants at an attendant console can use the Selective Conference Party Display, Drop, and Mute capability to display information about the participants on a call. When the system displays the information about a participant on a call, a user can drop a participant from the conference call or place a participant on mute.

To display the participants on a conference call:

- 1 Press the conference display feature button to place the station or the console in the conference display mode.

- 2 Press the conference display feature button repeatedly to scroll through the telephone numbers and names of each participant on the call. The telephone number of a participant is always available. The name of a participant might be unavailable.
- 3 To drop the participant that the system displays, press the Drop button.
This action is useful during a conference call when a user tries to add a participant that does not answer and the call goes to voice mail.
- 4 To place the participant that the system displays on mute, press the fe-mute button.
The remaining participants on the conference call cannot hear the participant who is on mute. A user or attendant can use the fe-mute button to place only one participant on mute, and that participant must be on a trunk call.
This ability to place a conference participant on hold is useful during conference calls when a participant puts the conference call on hold and the system plays music-on-hold to the remaining participants on the conference call.



CAUTION:

If a user repeatedly scrolls through the display information quickly, the system might take the telephone out of service. If the system takes the telephone out of service, the system resets the telephone, and drops the user from the call.

Reports for Conference

The following reports provide information about the Conference feature:

- None

Considerations for Conference

This section provides information about how the Conference feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Conference under all conditions. The following considerations apply to Conference:

- Trunk-to-Trunk Connections

If you do not allow trunk-to-trunk connections on your system, the system drops all conference participants when:

- A user hangs up.
- All the other participant connect to the conference through trunk lines.

- Loss Plan on Conference Calls

The end-to-end total loss for multiparty conference calls that is administered on the **Location Parameters** screen is not always applied to a specific call. The loss applied to a three-party conference call, for example, is calculated by adding the fixed pairwise loss for each pair of ports to the value for two-party loss shown on the **Location Parameters** screen. If this total is less than the end-to-end total loss value configured for a three-party conference, calculate the difference, and divide the difference by 2. Add 1 to this figure, and the result is the amount of loss applied to the call.

Interactions for Conference

This section provides information about how the Conference feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Conference in any feature configuration.

- Bridged Call Appearance

A user can use a bridged call appearance to make a conference call.

A bridged appearance can bridge onto a conference call if that action does not cause the number of participants on the conference call to exceed six participants.

- Call Vectoring

A call to a Vector Directory Number (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.

- Call Waiting

When a user on an analog single-line telephone activates the Call Wait feature, and the user creates a conference call, the Call Wait feature does not function while the user is on the conference call.

- Class of Restriction (COR)

If the **Restriction Override** field on the ***Class of Restriction*** screen is set to **all**, the system compares the COR of the participant who controls the conference call with the COR of a new conference participant. The system compares the COR of the participant who controls the conference call with the COR of a new conference participant before the system adds the participant to the conference call. The system does not compare the COR of the new participant with the CORs of the other conference participants.

If the COR of the user who controls the conference call allows the override of inward call restrictions, the system compares the COR of the user who controls the conference call with the COR of a new conference participant. The system compares the COR of the user who controls the conference call with the COR of a new conference participant before the system adds the new participant to the conference call. The system does not compare the COR of the new participant with the CORs of the other conference participants.

- Emergency Access to an Attendant

A user cannot make an Emergency-Access-to-an-Attendant call a participant in conference call.

- Trunk-to-Trunk Transfer

The system does not recognize the Conference button or the Transfer button when a user dials enough digits for the system to route a call, and the system can route the call differently if the user dials more digits. The user must be a user of a multifunction telephone, for example a BRI telephone, a digital telephone, or a hybrid telephone.

If the user wants the system to route the call based on the digits that the user has already dialed, the user must not dial any digits for three seconds, or the user must dial a pound sign (#). The system then recognizes the Conference button or the Transfer button and completes the call.

- VDN in a Coverage Path

Calls in an established conference do not cover to a vector directory number (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.

Data Call Setup

Use the Data Call Setup feature to set up a data call by any of following methods:

- Data-terminal or keyboard dialing
- Telephone dialing
- Hayes AT command dialing
- Permanent-switched connections
- Administered connections
- Automatic-calling unit interface (MPD and HSC)
- Hotline dialing

Detailed description of Data Call Setup

This section provides a detailed description of the Data Call Setup feature.

In addition to data-terminal and telephone dialing, the system accepts calls from other devices. For example, you can use a modular-processor data module (MPDM) that is equipped with an automatic-calling unit (ACU) interface module to dial from a host computer.

In addition to the numbers, the pound key (#), and the asterisk key (*) on a telephone. [Table 50, Special characters](#), on page 575 shows the special characters that a user can dial:

Table 50: Special characters

Character	Short description	Long description
SPACE or minus (-)	and	A space or minus (-) improves legibility. The server ignores these characters during dialing.
plus (+)	wait	A plus (+) interrupts or suspends dialing until the user receives dial tone.
comma (,)	pause	A comma (,) inserts a 1.5-second delay.
percent (%)	mark	Use a percent (%) to indicate that the digits are for end-to-end signaling when you use a touchtone trunk. Use a percent (%) with a rotary trunk. The percent (%) is not required when you use a touchtone trunk.
UNDERLINE or BACKSPACE	—	Use an underline or a backspace to correct characters that you typed on the same line.
at (@)	—	Use an at (@) to delete the entire line and start again with a new DIAL: prompt.

Each line of information that a user dials can contain 42 characters. Note that the system counts the plus (+) and the percent (%) as two characters each.

You administer the asynchronous data module (ADM) as one endpoint of a connection. The server establishes the connection at the scheduled time, and maintains the connection for the specified duration. After the call is accepted, the data set enters into continuous mode for the specified duration. If the server reboots during the connection, or if the connection drops, the server starts the connection again.

The system handles all ISDN basic rate interface (BRI) bearer data-call requests that are presently defined. If the server does not support a capability, the system returns a proper cause value to the terminal.

The system sends a cause code, also called a reason code, to BRI terminals to identify the reason that the system clears a call. The BRI data module converts some cause values to text messages for the system to display. In a passive-bus multipoint configuration, the system supports two BRI endpoints per port, and thus doubles the capacity of the BRI circuit pack. When you change the configuration of a BRI endpoint from point-to-point configuration to a multipoint configuration, the original endpoint does not need to reinitialize. In a multipoint configuration, you can administer only endpoints that support service profile identifier (SPID) initialization.

[Table 51, Call progress messages](#), on page 576 shows the call progress messages and the call progress descriptions for digital communication protocol (DCP) and ISDN-BRI modules that the system provides.

Table 51: Call progress messages

Message Code	Module Type	Message Description
DIAL:	DCP	This message is the equivalent of a dial tone. Enter the desired number or Feature Access Code (FAC), and then Press Enter .
CMD	BRI	This message is the equivalent of a dial tone. Enter the desired number or FAC, and then Press Enter .
RINGING	DCP, BRI	This message is the equivalent of a ringing tone. The called terminal is ringing.
BUSY	DCP, BRI	This message is the equivalent of a busy tone. The called number is busy or out of service.
ANSWERED	DCP, BRI	The call is answered.
ANSWERED-NOT DATA	DCP	The call is answered and the system does not detect a modem answer tone.
TRY AGAIN	DCP, BRI	This message is the equivalent of a reorder tone. The system facilities are unavailable.
DENIED	DCP, BRI	This message is the equivalent of an intercept tone. The system cannot place the call as dialed.
ABANDONED	DCP, BRI	The calling user abandoned the call.
NO TONE	DCP, BRI	The system does not detect a tone.

1 of 2

Table 51: Call progress messages

Message Code	Module Type	Message Description
CHECK OPTIONS	DCP, BRI	The data-module options are incompatible.
XX IN QUEUE	DCP, BRI	XX represents the position of the call in the queue.
PROCESSING	DCP, BRI	The call is out of the queue. The facility is available.
TIMEOUT	DCP, BRI	The call exceeds the time allowed. The system terminates the call.
FORWARDED	DCP, BRI	This message is the equivalent of a redirection-notification signal. The called terminal activates Call Forwarding and receives a call, and the system then forwards the call.
INCOMING CALL	DCP, BRI	This message is the equivalent of ringing.
INVALID ADDRESS	DCP	The user entered a name that is not in the defined in the Alphanumeric Dialing feature.
WRONG ADDRESS	BRI	The user entered a name that is not defined in the Alphanumeric Dialing feature.
PLEASE ANS-	DCP, BRI	The originating telephone user used the One-Button Transfer to Data capability to transfer the call to a data module.
TRANSFER	DCP	Data Call Return-to-Voice is occurring.
CONFIRMED	DCP, BRI	This message is the equivalent of the confirmation tone. The system either accepts the feature request of the user, or the system sends the call to a local coverage point.
OTHER END	DCP, BRI	The endpoint terminates the call.
DISCONNECTED	DCP, BRI	The system disconnects the call.
WAIT	DCP, BRI	The normal process continues.
WAIT, XX IN QUEUE	DCP	The call is in a local hunt-group queue.

2 of 2

The following data functions are unavailable on ISDN-BRI telephones:

- One-Button Transfer to Data
- Return-to-voice
- Data call preindication
- Voice-call transfer to data
- Data-call transfer to voice

Hardware requirements for Data Call Setup

The Data Call Setup feature requires the following hardware:

- None

Administering Data Call Setup

The following steps are part of the administration process for the Data Call Setup feature:

- [Defining a data module](#)
- [Specifying the port location](#)
- [Assigning the data extension feature button](#)

This section describes:

- Any prerequisites for administering the Data Call Setup feature
- The screens that you use to administer the Data Call Setup feature
- Complete administration procedures for the Data Call Setup feature

Prerequisites

You must complete the following actions before you can administer the Data Call Setup feature:

- Ensure that the feature access code (FAC) for data origination is available on your system.

To ensure that the FAC for data origination is available on your system:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the *Feature Access Codes (FAC)* screen ([Figure 146, Feature Access Code \(FAC\) screen](#), on page 579).

Figure 146: Feature Access Code (FAC) screen

change feature-access-codes FEATURE ACCESS CODE (FAC) Data Origination Access Code: Data Privacy Access Code: Directed Call Pickup Access Code: Emergency Access to Attendant Access Code: Enhanced EC500 Activation: *45 Deactivation: *46 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: *03 Leave Word Calling Message Retrieval Lock: *33 Leave Word Calling Message Retrieval Unlock: *34 Leave Word Calling Send A Message: *35 Leave Word Calling Cancel A Message: *37 Malicious Call Trace Activation: Deactivation: Meet-me Conference Access Code Change:	Page 2 of 7

- 2 Click **Next** until you see the Data Origination Access Code field.
- 3 In the Data Origination Access Code field, type the FAC for data origination access.
- 4 Press **Enter** to save your changes.

Screens for administering Data Call Setup

Table 52: Screens for data-terminal dialing

Screen name	Purpose	Fields
Data Module PDM/TDM Data Line 7500	Define the data module.	- All - All - All
Modem Pool Group	Specify the port that is associated with the conversion resource on the integrated modem pool circuit pack.	Circuit Pack Assignments

Table 53: Screens for telephone dialing

Screen name	Purpose	Fields
<i>Feature Access Codes</i>	Assign the Feature Access Code (FAC) for data origination.	Data Origination Access Code
<i>Station</i>	Assign the feature button for data extension.	data-ext (Ext:)
<i>Data Module</i> • <i>PDM/TDM</i> • <i>Data Line</i>	Define the data module.	• All • All
<i>Modem Pool Group</i>	Specify the port that is associated with the conversion resource on the integrated modem pool circuit pack.	Circuit Pack Assignments

Defining a data module

To define a data module:

- 1 Type **add data-module next**. Press **Enter**.

The system displays the **Data Module** screen ([Figure 147, Data Module screen - pdm data module](#), on page 580).

Figure 147: Data Module screen - pdm data module

add data-module next		Page 1 of 1	
DATA MODULE			
Data Extension: 502	Name:	BCC: 2	
Type: pdm	COS: 1	Remote Loop-Around Test? n	
Port:	COR: 1	Secondary data module? n	
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 (first 26 characters)		
1:			

Figure 148: Data Module screen - 7500 data module

add data-module next		Page 1 of 2
	DATA MODULE	
Data Extension: 502	Name:	
Type: 7500	COS: 1	
Port:	COR: 1	Multimedia? n
	TN: 1	
ABBREVIATED DIALING		
List1: group 1		
SPECIAL DIALING OPTION:		
CIRCUIT SWITCHED DATA ATTRIBUTES		
Default Duplex: full	Default Mode: async	Default Speed: 1200
DATA MODULE CAPABILITIES		
Default ITC: restricted		Default Data Application: M2_A

Figure 149: Data Module screen - 7500 data module

add data-module next		Page 2 of 2
	DATA MODULE	
BRI LINK/MAINTENANCE PARAMETERS		
XID? y	Fixed TEI? n	
MIM Support? y	Endpt Init? y	SPID: 502
		MIM Mtce/Mgt? y

Figure 150: Data Module screen - data-line data module

add data-module next		Page 1 of 2	
DATA MODULE			
Data Extension: 502	Name:	BCC: 2	
Type: data-line	COS: 1		
Port:	COR: 1		
ITC: restricted	TN: 1	Connected To: dte	
ABBREVIATED DIALING			
List1: personal 1			
SPECIAL DIALING OPTION:			
ASSIGNED MEMBER (Station with a data extension button for this data module)			
Ext	Name (first 26 characters)		
1:			

Figure 151: Data Module screen - data-line data module

add data-module next		Page 2 of 2	
DATA MODULE			
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	

The system displays two display-only fields in the **ABBREVIATED DIALING** area. These display only-fields are **EXT** and **Name**. The fields contain the extension number and the name of the users who have associated data extension buttons, and who share this data module.

- 2** In the **BCC** field, type a **1** if the speed is 56 kbps, or type either a **2**, **3**, or **4** if the speed is 64 kbps.

The system compares the speed setting that you assign here with the speed setting in an associated routing pattern. The system compares the two speed settings when calls that attempt to use the data module fail to complete.

The system displays the **BCC** field appears if the **ISDN-PRI** field or the **ISDN-PRI Trunks** field on the **Optional Features** screen is set to **y**.

- 3** The **CAPABILITIES** area contains three fields.

- In the **Busy Out** field, perform one of the following actions:
 - Type **y** to place the data line circuit (DLC) port in a busy-out state so calls do not terminate at the data terminal equipment when the DTE control lead to the DLC drops. Use this option for DTEs that are members of a hunt group.
 - Type **n** to keep the DCL port out of a busy-out state when the DTE control lead to the DLC drops.
- The system displays the **Configuration** field only when the **KYBD Dialing** field is set to **y**.
 - Type **y** if you want to view and change options from originating or receiving DTEs, such as non intelligent terminals.
 - Type **n** if you do not want view and change options from intelligent devices such as computers.
- In the **KYBD Dialing** field, perform one of the following actions:
 - Type **y** if you want the users to dial calls from a keyboard, and to allow the data module endpoint to transmit and receive text during call origination or call termination.

If you type **y**, you must also type **n** in the **Low** field in the **SPEEDS** area of the **Data Module** screen.
 - Type an **n** if you do not want the users to dial calls from a keyboard. If you type **n**, the data module endpoint cannot transmit and receive text during call origination and call termination. If you type an **n**, data calls can be answered, but there is no text feedback.

- 4** The **CIRCUIT SWITCHED DATA ATTRIBUTES** area contains information that is used with 7500 data modules and World Class BRI data modules.

Note that the fields in the **CIRCUIT SWITCHED DATA ATTRIBUTES** area contain default information. The default information is for modem pooling conversion resource insertion when the endpoint does not support the data query capability, and for when the endpoint does not support the administered connections. The information in the fields has no significance for data modules that provide data query, such as Avaya-supported ISDN-BRI data modules. Use the system default settings that the system provides for Avaya ISDN-BRI data modules or World Class ISDN-BRI data modules.

- In the **Default Duplex** field, perform one of the following actions:
 - Type **full** to allow simultaneous, two-way transmission.

This is duplex mode.
 - Type **half** to allow only one transmission direction at a time.

This is half duplex mode.

- In the **Default Mode** field, perform one of the following actions:
 - Type **sync** for synchronous data mode.
 - Type **async** for asynchronous data mode.
 - In the **Default Speed** field, type the data rate. The valid entries are:
 - **1200**
 - **2400**
 - **4800**
 - **9600**
 - **19200**
 - **56000** when the **Default Mode** field is set to **sync**
 - **64000** when the **Default Mode** field is set to **sync**
- 5 The system displays the **Connected To** field, when the **Type** field contains either **dpm** or **data-line**.
- In the **Connected To** field, perform one of the following actions:
- Type **dte** if the Asynchronous Data Unit (ADU) is connected to DTE.
 - Type **isn** if the ADU is connected to an information systems network.
- 6 In the **COR** field, type the number of the class of restriction for this data module. Valid entries are **0** through **95**.
- 7 In the **COS** field, type the number of the class of service for this data module. Valid entries are **1** through **15**.
- 8 The **DATA MODULE CAPABILITIES** area contains three fields with information for the 7500 data modules and the World Class BRI (WCBRI) data modules.
- The **Default Data Applications** field identifies the mode that the system uses to originate calls when the calling parameters do not specify the mode. The system also uses the mode to terminate trunk calls that do not have administered connections or for which the bearer capability is unspecified. See the “Uniform Dial Plan feature” for more information.
 - In the **Default Data Applications** field, perform one of the following actions:
 - Type **M0** to specify mode 0. Use this option for a WCBRI endpoint that the system uses as an administered connection.
 - Type **M1** to specify mode 1.
 - Type **M2_A** to specify mode 2 asynchronous.
 - Type **M2_S** to specify mode 2 synchronous.
 - Type **M3/2** to specify mode 3/2 adaptable.
 - In the **Default ITC** field, perform one of the following actions:
 - Type **restricted** for a WCBRI endpoint that is an administered connection.
 - Type **unrestricted** for a WCBRI endpoint that is not an administered connection.
 - The display-only **MM Complex Voice Ext** field contains the number of the associated phone in the multimedia complex. The system displays the **MM Complex Voice Ext** field only when the **Multimedia** field is set to **y**.

The field is blank until you type the data module extension in the MM Complex Data Ext field on the **Station** screen. When you type the data module extension in the MM Complex Data Ext field on the **Station** screen, the system associates the numbers in the MM Complex Data Ext and the MM complex Voice Ext fields as two parts of a one-number complex. The one-number complex is the extension of the telephone.

The system displays the data module extension in the display-only Data Extension field.

- 9** The system does not display the ITC field for voice-only stations or for BRI stations. The ITC field applies only when the Comm Type field on the **Trunk Group** screen, that the system uses for an outbound call, contains **avd** or **rbavd**. The ITC field specifies the type of transmission facilities that an ISDN call uses when a call originates from this data module endpoint.

In the ITC field, perform one of the following actions:

- Type **restricted** if the data module can send bits at speeds less than or equal to 56 kbps. If you type **restricted** in the ITC field, the system uses a trunk group for which the COMM Type field on the **Trunk Group** screen is set to **rbavd** or to **avd** to complete a call from this data module endpoint.

A restricted transmission facility enforces ones density digital transmission. Ones density digital transmission is a sequence of eight digital zeroes that the firmware on the DS1 port board converts to a sequence of seven zeroes and a digital 1.

- Type **unrestricted** if the data module can send bits at a speed no greater than 64 kbps. If you type **unrestricted** in the ITC field, the system uses a trunk group for which the COMM Type field on the **Trunk Group** screen is set to **avd** to complete a call from this data module endpoint. The value **avd** in the Comm Type field indicates that the trunk group provides both restricted and unrestricted transmission facilities.

An unrestricted transmission facility does not enforce ones density digital transmission. The DS1 port board firmware does not convert the digital information.

- 10** In the List1 field in the ABBREVIATED DIALING area, perform one of the following actions:

- Leave the field blank if you do not want the data module to have an abbreviated dialing list.
- Type **e** if you want the data module to have an enhanced abbreviated dialing list.
- Type **g** if you want the data module to have a group list. If you type **g**, the system displays a field to the right of the List1 field. If you type **g** in the List1 field, you must also type a group list number in the field that the system displays.
- Type **p** if you want the data module to have a personal list. If you type a **p**, the system displays a field to the right of the List1 field. If you type a **p** in the List1 field, you must type a personal list number in the field that the system displays.
- Type **s** if you want the data module to have a system abbreviated dialing list.

- 11** In the Name field, perform one of the following actions:

- Type the name of the user who is associated with the data module.
- Leave the field blank.

12 The **OPTIONS** area contains six fields.

- The system displays the **Answer Text** field only if the **KBDY Dialing** field is set to **y**.

The **Answer Text** field applies to the following call messages:

- Incoming
- Answered
- Disconnected
- Disconnected other end

In the **Answer Text** field, perform one of the following actions:

- Type **y** to allow text feedback to the DTE when a user answers a call or the system disconnects a call. The text feedback includes both DLC-generated text and system-generated text.
 - Type **n** to disable text feedback to the DTE when a user answers a call or the system disconnects a call, and when the DTE that answers a call is a computer or an intelligent device. The system still generates the text, but the DLC does not allow the text to be delivered to the DTE.
- The system displays the **Connected Indication** field only if the **KBDY Dialing** field is set to **y**. If the **Connected Indication** field is set to **n**, DLC provides the connection indication when the DLC activates the Electronics Industries Association (EIA) 232C control lead.

In the **Connected Indication** field, perform one of the following actions:

- Type **y** if you want the system to generate a “connected” message to the DTE when the system establishes a connection.
 - Type **n** if you do not want the system to generate a “connected” message to the DTE when the system establishes a connection.
- The system displays the **Dial Echoing** field only if the **KBDY Dialing** field is set to **y**.

In the **Dial Echoing** field, perform one of the following actions:

- Type **y** if you want the system to echo characters back to the DTE.
 - Type **n** if you do not want the system to echo characters back to the DTE and when an intelligent device provides keyboard dialing.
- The system displays the **Disconnect Sequence** field only if the **KBDY Dialing** field is set to **y**.

In the **Disconnect Sequence** field, perform one of the following actions:

- Type **long-break** if you want a break that is greater than 2 seconds.
- Type **two-breaks** if you want a break that is less than 1 second.

- The system displays the **Parity** field only if the **KBDY Dialing** field is set to **y**. The DLC generates the parities when the DLC sends call setup text to the DTE. The DLC does not check the parity when the DLC receives dial characters. Select the parity that matches the DTE that connects to the data module.

In the **Parity** field, type one of the following types of parity:

- **even**
- **odd**
- **mark**
- **space**

- The **Permit Mismatch** field contains information that allows an EIA interface to operate at a rate that differs from the rate that is agreed upon during the data module handshake. The rate that is agreed upon during the data module handshake is always the highest compatible rate among the speeds that each data module reports.

The information in the **Permit Mismatch** field eliminates the need to change the DTE or DLC speed whenever someone, or something, places a call to or from endpoints that operate at a different speed.

When the **Permit Mismatch** field is set to **y**, the DLC reports the highest optional speed and all the lower speeds, or the previously selected autoadjust speed, during the handshake process.

- Type **y** if you want the DLC to operate at the highest selected speed, which is a higher rate than the far-end data module.
- Type **n** if you do not want DLC to operate at the highest selected speed.

13 In the **Port** field, type the appropriate values from [Table 54, Port field values](#), on page 587.

Table 54: Port field values

Characters	Description	Value
1-2	Cabinet Number	01 through 44 (For DEFINITY R configurations) 01 through 03 (For DEFINITY SI configurations) 01 through 64 (For S8700 IP-Connect)
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 SI, S8700 IP-Connect (tdm, pdm) configurations) 01 through 16 (ppp for S8700 IP-Connect) 01 through 08 (system-port for S8700 IP-Connect) 17/33 (ethernet on S8700 IP-Connect)
x	Administration without Hardware	If the Secondary data module? field, contains an n , you can type x in the Port field. A Port field set to x indicates that no hardware is associated with the port assignment.

14 The **SPEEDS** area contains information about the operating speeds of the data module.

- In the **Low** field, perform one of the following actions:
 - Type **y** if you want the data line circuit to operate at a speed of 0 to 1800 bps.
 - Type **n** if the **KYBD Dialing** field in the **CAPABILITIES** area is set to **y**.
- In the **300**, **1200**, **2400**, **4800**, **9600**, and **19200** fields, perform one of the following actions:
 - Type **y** if you want the DLC to operate at the speed.

You can choose any of the speeds for the DLC. The DLC matches the speed for the duration of the call.

If you select multiple speeds, you must also set the **Autoadjust** field to **n** and select at least three speeds. The speed of the DTE must be the highest speed that you select. The DTE must have the highest speed because the system delivers feedback to the DTE at the highest selected speed.

— Type **n** if you do not want the DLC to operate at the speed.

- The system displays the **Autoadjust** field when the **KYBD Dialing** field in the **CAPABILITIES** area is set to **y**. The **Autoadjust** field applies only to calls that a user originates from a keyboard.

In the **Autoadjust** field, perform one of the following actions:

— Type **y** if you want the DLC port to automatically adjust to the operating speed and the parity of the DTE to which the DLC port connects.

— Type **n** if you do not want the DLC port to automatically adjust to the operating speed and the parity of the DTE to which the DLC port connects.

- 15** In the **SPECIAL DIALING OPTION** field, perform one of the following actions:

- Leave the field blank if you do not want the data module to have special dialing
- Type **hot-line**, if you want the data module to have hot-line dialing. If you type **hot-line**, the system displays the **Abbreviated Dialing Dial Code (from above list):** field. Type the **abbreviated dial code** in the **Abbreviated Dialing Dial Code (from above list):** field. Valid entries are **0** through **999**.
- Type **default**, if you want the data module to have default dialing. If you type **default**, the system displays the **Abbreviated Dialing Dial Code (from above list):** field. Type the **abbreviated dial code** in the **Abbreviated Dialing Dial Code (from above list):** field. Valid entries are **0** through **999**.

- 16** In the **TN** field, type the tenant partition number of the data module. Valid entries are **1** through **100**.

- 17** In the **Type** field, perform one of the following actions:

- Type **7500** to assign a 7500 data module.

The 7500 data module supports:

- Automatic TEI
- B-channel, maintenance and management messaging
- Service Profile Identifier (SPID) initialization capabilities.

BRI voice endpoints, BRI data endpoint, or both BRI voice and BRI data endpoints 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 circuit pack can support up to 12 ports.

You can administer more than one ISDN endpoint, either a voice endpoint or a data endpoint, on one port. You can administer more than one ISDN endpoint, because BRI provides a multipoint capability.

For BRI, multipoint administration allows for telephones that have SPID initialization capabilities. Multipoint administration is allowed only if no endpoint that is administered on the same port is a fixed tie endpoint, and no station on the same port has B-channel data capability. The system restricts multipoint administration to two endpoints per port.

- Type **data-line** to assign a data line data module.

Use the **Data Line Data Module (DLDM)** screen to assign ports on the Data Line (DLC) circuit pack that allow EIA 232C devices to connect to the system. The DLC, with a companion 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 of 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 the wire gauge.

The DLC has eight ports. The connection from the port to the EIA device is *direct*, which means 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 DTE, and as a digital line port to the server that runs Avaya Communication Manager.

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

- Type **pdm** to assigns a DCE interface for processor/trunk data modules.

Use these screens assign Modular Processor Data Modules (MPDMs) and Modular Trunk Data Modules (MTDMs). Use one screen to assign MPDMs (700D), 7400B, 7400D or 8400B Data Module. Use another screen for MTDMs (700B, 700C, 700E, 7400A). You must complete one screen for each MPDM, 7400B, 7400D, 8400B or MTDM.

The MPDM, 7400B, or 8400B Data Module provides a Data Communications Equipment (DCE) interface. Use the interface for a connection to equipment such as a data terminal, call detail recording (CDR) output device, on-premises administration terminal, Message Server, Property Management System (PMS), AUDIX, and host computers. The MPDM, 7400B, or 8400B Data Module also provides a Digital Communications Protocol (DCP) interface to the digital switch. Note that DCE is the equipment on the network side of a communications link that provides all the functions that are required to make the binary serial data from the source or transmitter compatible with the communications channel.

The MTDM provides an EIA 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. Note that DTE is the equipment that comprises 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 the 7400A Data Module can also serve as part of a conversion resource for combined modem pooling.

- 18** Press **Enter** to save your changes.

Specifying the port location

To specify the port location:

- 1 Type **change modem-pool num** *n*, where *n* is the modem pool that you want to change. Press **Enter**.

The system displays the **Modem Pool Group** screen ([Figure 152, Modem Pool Group screen](#), on page 590).

Figure 152: Modem Pool Group screen

change modem pool group		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 Location	Circuit Pack Location
	1: ____	9: ____
	2: ____	10: ____
	3: ____	11: ____
	4: ____	12: ____
	5: ____	13: ____
	6: ____	14: ____
	7: ____	15: ____
	8: ____	16: ____

- 2 In the **Circuit Pack Location** field, type the 7-character port number that is associated with the conversion resource on the integrated modem pool circuit pack.

Information in the **Circuit Pack Location** field is optional for integrated conversion resources only.

- 3 Press **Enter** to save your changes.

Assigning the data extension feature button

To assign the data extension feature button:

- 1 Type **change station n**, where *n* is the telephone number of the user that you want to change. Press **Enter**.

The system displays the **Station** screen ([Figure 153, Station screen](#), on page 591).

Figure 153: Station screen

change station 32023		Page 3 of 3
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: data-ext	Ext: 22000
2: call-appr	5:	
3: call-appr	6:	

- 2 Page through the screens until you find the **BUTTON ASSIGNMENTS** area.
- 3 Move to the button number that you want to use for call pickup. You can use any of the buttons.
- 4 Type **data-ext** after the button number.
- 5 When you type **data-ext** a field, the system displays an **Ext :** field.
Type the extension of the data module in the **Ext :** field.
- 6 Press **Enter** to save your changes.

End-user procedures for Data Call Setup

End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Data Call Setup:

- [Setting up and disconnecting data calls from a DCP data terminal](#)
- [Setting up data calls from a DCP telephone](#)
- [Setting up and disconnecting data calls from an ISDN-BRI data terminal](#)
- [Setting up data calls from an ISDN-BRI telephone](#)

Setting up and disconnecting data calls from a DCP data terminal

To set up and disconnect data calls from a Digital Communications Protocol (DCP) data terminal.

- 1** At the Dial: prompt, type the data number.
The system displays the RINGING message.
If the call is in a queue, the system displays the WAIT, XX IN QUEUE message. The system displays the position of the call in the queue, represented by XX, as the system moves the call through the queue.
- 2** Press **Break** to originate or disconnect a call.
- 3** If the terminal does not generate a 2-second continuous break signal, press the **originate/disconnect** button on the data module.
- 4** At the DIAL: prompt, type the digits.

Setting up data calls from a DCP telephone

When a data terminal is unavailable, you can originate and control data calls from a DCP telephone. Use any unrestricted telephone to set up the call, and then transfer the call to a data module endpoint.

Use a button on a multiappearance telephone data-extensions to make the data call. Assign any administrable feature button as a data-extension button. The data-extension button provides one-touch access to a data module.

Use any of the following options, either alone or in combination, to make a data call from a voice terminal.

- **One-Button Transfer to Data**
Press the **data-extension** button after the endpoint answers, to transfer a call to the associated data module.
- **Return-to-Voice**
Press the **data-extension** button that is associated with a busy data module to change a connection from a data connection to a voice connection. If you hang up, the system disconnects the call. If the system returns the data call to the telephone, the system either continues the call in voice mode, or the system transfers the data call to another endpoint.
- **Data Call Preindication**
Press the data-extension button to reserve the associated data module before you dial a data endpoint. The system reserves the data module for the call and reserves a conversion resource for the call, if the call needs a conversion resource.
Use the Data Call Preindication option before you use one-button transfer to data, for data calls that use toll-network facilities. Data Call Preindication is in effect until you press the associated **data-extension** button again for a one-button transfer. There is no timeout.

Setting up and disconnecting data calls from an ISDN-BRI data terminal

To set up and disconnect data calls from an ISDN-BRI data terminal:

- 1** Press **Enter** a few times until the system displays the CMD: prompt.
- 2** If the system does not display the CMD: prompt:
 - a** Press **Break**, **A**, and **T** at the same time.
 - b** Press **Enter**.
 - c** Type three pluses (+++).
The system displays the CMD: prompt.
 - d** Type **end**.
 - e** Press **Enter**.

Setting up data calls from an ISDN-BRI telephone

To set up a data call from an ISDN-BRI telephone:

- 1** Press the data button on the telephone.
- 2** Type the number on the dial pad.
- 3** Press the data button again.

The following data functions are unavailable on ISDN-BRI voice terminals:

- One-button transfer to data
- Return-to-voice
- Data call preindication
- Voice call transfer to data, and data call transfer to voice

Reports for Data Call Setup

The following reports provide information about the Data Call Setup feature:

- None

Considerations for Data Call Setup

This section provides information about how the Data Call Setup feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Data Call Setup under all conditions. The following considerations apply to Data Call Setup:

- ISDN basic rate interface (BRI) has a voice-to-data restriction. A telephone cannot call a data terminal, and a data terminal cannot call a telephone
- BRI telephones cannot have data-extension buttons. Digital Communications protocol (DCP) sets have data-extension buttons. However, DCP sets cannot have data-extension buttons for BRI.
- When a telephone user uses a modem to place a data call, the user dials the data-origination access code that is assigned in the system before the user dials the endpoint.
- The system does not limit the number of assigned data-extension buttons per telephone. Assign telephone buttons that access the data module.
- Telephone dialing is not available in ISDN-BRI applications because ISDN-BRI terminals supports nether voice-call transfer to data, nor data-call transfer to voice.

Interactions for Data Call Setup

This section provides information about how the Data Call Setup feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Data Call Setup in any feature configuration.

- **Abbreviated Dialing**
You can use only 22 of the 24 digits in an abbreviated-dialing number when you dial at a keyboard. The remaining two digits must contain the wait indicator for tone detection.
- **Alphanumerical Dialing**
When a data-terminal user uses the Alphanumeric Dialing feature to place a call, the user enters an alphanumeric name.
- **Call Coverage**
You cannot assign a coverage path to a hunt group that consists of data endpoints.
- **Call Detail Recording (CDR)**
CDR records the use of modem pools on trunk calls.
- **Call Forwarding All Calls**
The system uses the Call Forwarding All Calls capability to redirect data calls to a user-designated extension. The attendant or a forwarding party dials an FAC to activate the Call Forwarding All Calls capability.

A user can use data-terminal dialing to activate Call Forwarding All Calls for calls to a data module. If the forwarded-to endpoint is an analog endpoint, and the caller is a digital endpoint, the system activates modem pooling automatically.

- **Data Hotline**

Data Hotline is a security feature. The server terminates calls to a preadministered hotline. The system discards any address string and routes the call as if the users entered the hotline-destination address. This Data Hotline feature does not affect incoming calls. You cannot use the Data Hotline feature and the Default Dialing feature at the same time.
- **Default Dialing**

If the Default Dialing feature is active, a data-terminal user can Press **Enter** to call a preadministered destination. The data-terminal user enters a complete address to call other destinations.
- **Digit Dialing**

The system provides basic digit dialing through an asynchronous data module (ADM) or a 7500B data module. The user can enter digits from 0 to 9, an asterisk (*), and a pound sign (#) from a 7500 series telephone keypad or from an Electronics Industries Association (EIA) terminal interface.
- **Internal Automatic Answer**

Data calls are not eligible for Internal Automatic Answer.
- **Modem Pooling**

Modem Pooling is available on data calls. The system automatically inserts a modem if the data call needs a modem. You can use the Data Call Preindication option or the Data Origination option to indicate the need for a modem.
- **Uniform Call Distribution (UCD)**

UCD provides a group of data modules, or analog modems, for answering calls to a connected facility. A computer port is an example of a connected facility.
- **World-Class Tone Detection**

The system supports multiline data-terminal dialing, if you set the **tone-detection options** field on the ***Feature-Related System-Parameters*** screen to **precise**.

The message that Data Call Setup sends to users depends on the tone-detection option that you administer.

Default Dialing

Use the Default Dialing feature to provide data-terminal users who often dial the same number a very simple method to dial that number. Normal data-terminal dialing and the alphanumeric dialing features are unaffected.

Detailed description of Default Dialing

This section provides a detailed description of the Default Dialing feature.

Data terminal users use the computer keyboard to dial. With Default Dialing, a data-terminal user can place a data call to a preadministered destination by doing either of the following:

- Press **Enter** at the **DIAL: prompt** (for data terminals using DCP 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).
- Type **d** and press **Enter** at the **CMD: prompt**. For data terminals that use ISDN-BRI data modules.
To place calls to other destinations, the user calls types **d**, a space, the complete address, and press **Enter** after the **CMD: prompt**.

NOTE:

DU-type hunt groups that connect the system to a terminal server on a host computer have hunt-group extensions set to **no** keyboard dialing.

Hardware requirements for Default Dialing

The Default Dialing feature requires the following hardware:

- None

Administering Default Dialing

This section describes the screens that you use to administer the Default Dialing feature.

Screens for administering Default Dialing

Screen name	Purpose	Fields
<i>Data Module</i>	Set the default dialing options.	• Special Dialing Option • Abbreviated Dialing List • AD Dial Code

Reports for Default Dialing

The following reports provide information about the Default Dialing feature:

- None

Considerations for Default Dialing

This section provides information about how the Default Dialing feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Default Dialing under all conditions. The following considerations apply to Default Dialing:

- None

Interactions for Default Dialing

This section provides information about how the Default Dialing feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Default Dialing in any feature configuration.

- None

Demand Print

Use the Demand Print feature to print undelivered messages.

Detailed description of Demand Print

This section provides a detailed description of the Demand Print feature.

With Demand Print, a user can enter a feature access code (FAC) or press a button to print undelivered messages.

Hardware requirements for Demand Print

The Demand Print feature requires the following hardware:

- A printer

Administering Demand Print

This section describes the screens that you use to administer the Demand Print feature.

Screens for administering Demand Print

Screen name	Purpose	Fields
<i>Feature Access Code (FAC)</i>	Specify the FAC for Demand Print.	Print Messages Access Codes
<i>Station (multiappearance)</i>	Assign a Demand Print feature button for a user.	Button/Feature Button Assignments • print-msgs
	Assign a station security code (SCC) for a user.	Security Code

Reports for Demand Print

The following reports provide information about the Demand Print feature:

- None

Considerations for Demand Print

This section provides information about how the Demand Print feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Demand Print under all conditions. The following considerations apply to Demand Print:

- None

Interactions for Demand Print

This section provides information about how the Demand Print feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Demand Print in any feature configuration.

- None

Dial Access to Attendant

Use the Dial Access to Attendant feature to reach an attendant by dialing an attendant access code.

Detailed description of Dial Access to Attendant

The Dial Access to Attendant feature allows telephone users in your system to reach an attendant by dialing an attendant access code. Attendants can then extend the call to a trunk, or to another telephone.

Hardware requirements for Dial Access to Attendant

The Dial Access to Attendant feature requires the following hardware:

- None

Administering Dial Access to Attendant

The following steps are part of the administration process for the Dial Access to Attendant feature:

- [Changing the attendant access code](#)

This section describes:

- Any prerequisites for administering the Dial Access to Attendant feature
- The screens that you use to administer the Dial Access to Attendant feature
- Complete administration procedures for the Dial Access to Attendant feature

Prerequisites for administering Dial Access to Attendant

You must complete the following actions before you can administer the Dial Access to Attendant feature:

- None

Screens for administering Dial Access to Attendant

Screen name	Purpose	Fields
<i>Dial Plan Analysis Table</i>	Change the attendant access code from the default setting of 0 .	• Dialed String • Total Length

Changing the attendant access code

You can administer the length of the attendant access code. The attendant access code can be a 1-digit or a 2-digit number. The default attendant access code is **0**.

To change the attendant access code:

- 1 Type **change dialplan analysis**. Press **Enter**.

The system displays the *Dial Plan Analysis Table* screen ([Figure 154, Dial Plan Analysis Table screen](#), on page 602).

Figure 154: Dial Plan Analysis Table screen

change dialplan analysis				DIAL PLAN ANALYSIS TABLE				Page 1 of 12		
	Dialed String	Total Length	Call Type		Dialed String	Total Length	Call Type	Percent Full: 7		
	00	2	attd					Dialed String	Total Length	Call Type
	1	3	dac							
	2	4	ext							
	3	4	ext							
	3	1	aar							
	4	1	ars							
	4	5	ext							
	5	5	ext							
	5	7	ext							
	6	5	ext							
	7210	7	ext							
	8	7	ext							
	9	1	fac							
	*	3	fac							
	#	3	fac							

- 2 To assign a digit other than **0**, find **attd** in the **Call Type** column. In that row:
 - Change the number in the **Dialed String** column to a unique 1-digit or a 2-digit number.
 - If you changed the number in the **Dialed String** column to a 2-digit number, change the number in the **Total Length** column to **2**.
- 3 Press **Enter** to save your changes.

Starting with Communication Manager release 2.0, you can also enter a **fac** or **dac** entry on the **Dial Plan Analysis Table** screen to administer the attendant access code. You then enter the actual access code on the **Feature Access Codes** screen. You can administer location-specific attendant access codes on the **Locations** screen.

For more information, see the “Dial Plan” feature and the “Feature Access Code” feature.

Reports for Dial Access to Attendant

The following reports provide information about the Dial Access to Attendant feature:

- None

Considerations for Dial Access to Attendant

This section provides information about how the Dial Access to Attendant feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Dial Access to Attendant under all conditions. The following considerations apply to Dial Access to Attendant:

- None

Interactions for Dial Access to Attendant

This section provides information about how the Dial Access to Attendant feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Dial Access to Attendant in any feature configuration.

- Class of Restriction
If the Class of Restriction (COR) of a telephone restricts a user from originating calls, the user cannot call the attendant.
- Conference
If a telephone user dials the attendant access code, the attendant cannot add that user to an existing conference call.

Dial Plan

Use the Dial plan feature to understand how the Communication Manager software describes interprets dialed digits. For example, if you dial 9 on your system to access an outside line, the dial plan directs the system to find an external trunk when a dialed string begins with 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 a 1 or a 2.

All feature access codes, extensions and trunk access codes must be consistent with the dial plan.

Detailed Description of Dial Plan

This section describes the following topics:

- [Dial Plan Analysis Table](#)
- [Dial Plan Parameters](#)
- [Multi-Location Dial Plan](#)
- [Uniform Dial Plan \(UDP\)](#)

Dial Plan Analysis Table

The **Dial Plan Analysis Table** is the guide that the software uses to translate the digits dialed by users. This screen enables you to determine the beginning digits and total length for each type of call that Avaya Communication Manager needs to interpret.

Figure 1 shows an example of a simple dial plan ([Figure 155](#)).

Figure 155: Dial Plan Analysis Table screen

change dialplan analysis			DIAL PLAN ANALYSIS TABLE			Page 1 of 1		
						Percent Full: 9		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
0	1	attd	—	—	—	—	—	—
1	3	dac	—	—	—	—	—	—
3	5	ext	—	—	—	—	—	—
4	6	ext	—	—	—	—	—	—
5	7	ext	—	—	—	—	—	—
*	3	fac	—	—	—	—	—	—
#	3	fac	—	—	—	—	—	—
—	—	—	—	—	—	—	—	—
—	—	—	—	—	—	—	—	—
—	—	—	—	—	—	—	—	—
—	—	—	—	—	—	—	—	—

Use the **Dial Plan Analysis Table** screen to define the dial plan for your system.

- **Call Type** – Indicates what the system does when a user dials the digit or digits indicated in the **Dialed String** column. The **Dial Plan Analysis Table** screen 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 the 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 the example figure, dial access codes begin with the number **1** and are 3 digits long.
- NOTE:**
The **Dial Plan Analysis Table** screen 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** screen.
- **Extensions (ext)** – Defines extension ranges that can be used on your system.
In the example figure, extensions must be in the ranges 30000–39999, 40000–49999, and 50000–59999.
 - **Feature access codes (fac)** – 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 the example figure, feature access codes can begin with * or #, and are 3-digits long.
 - **Total Length** – Indicates how long the dialed string will be for each type of call.
In the example figure, the dial plan shows that when users dial a 5-digit number that starts with **3**, they are dialing an extension.

Dial Plan Parameters

The **Dial Plan Parameters** screen works with the **Dial Plan Analysis Table** to fully define the dial plan of your system. The **Dial Plan Parameters** screen allows you to set system-wide parameters for your dial plan. The **Dial Plan Parameters** screen also controls the appearance of 6-digit and 7-digit extensions on station displays. You can select a system-wide format to display all 6-digit extensions, and a format to display all 7-digit extensions.

The following figure shows a simple *Dial Plan Parameters* screen.

Figure 156: Dial Plan Parameters screen

change dialplan parameters	DIAL PLAN PARAMETERS	Page 1 of 1
Local Node Number: <u>2</u>		
ETA Node Number: <u>—</u>		
ETA Routing Pattern: <u>—</u>		
UDP Extension Search Order: <u>local-extensions-first</u>		
6-Digit Extension Display Format: <u>xx.xx.xx</u>		
7-Digit Extension Display Format: <u>xxx-xxxx</u>		
AAR/ARS Internal Call Prefix:		
AAR/ARS Internal Call Total Length:		

For more information on fields in the *Dial Plan Parameters* screen, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Multi-Location Dial Plan

When a customer migrates from a multiple independent node network to a single distributed server whose gateways are distributed across a data network, it may initially appear as if some dial plan functions are no longer available.

The multi-location dial plan feature preserves dial plan uniqueness for extensions and attendants that were provided in a multiple independent node network, but appear to be unavailable when customers migrate to a single distributed server. This feature is available beginning with Communication Manager, Release 2.0.

For example, in a department store with many locations, each location might have had its own switch with a multiple independent node network. The same extension could be used to represent a unique department in all stores (extension 4567 might be the luggage department). If the customer migrates to a single distributed server, a user could no longer dial 4567 to get the luggage department in their store. The user would have to dial the complete extension to connect to the proper department.

Instead of having to dial a complete extension, the multi-location dial plan feature allows a user to dial a shorter version of the extension. For example, a customer can continue to dial 4567 instead of having to dial 123-4567.

Communication Manager takes the location prefix and adds those digits to the front of the dialed number. The switch then analyzes the entire dialed string and routes the call based on the administration on the Dial Plan Parameters form.

Other options for 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.

Uniform Dial Plan (UDP)

The Uniform Dial Plan (UDP) provides the ability to share a common dial plan among a group of servers. The UDP applies both interserver dialing and intraserver dialing. The UDP provides the following types of dial plans:

- 3-digit dial plan
- 4-digit dial plan
- 5-digit dial plan
- 6-digit dial plan
- 7-digit dial plan
- A combination of any of these dial plans

UDP provides extension-to-extension dialing among two or more private-switching systems.

You can use UDP with the following entities:

- Main servers
- Tributary servers
- Satellite servers
- Electronic Tandem Networks (ETN)
- Distributed Communication Systems (DCS)

Note that you must use a 4-digit dial plan or a 5-digit dial plan for DCS.

See the Uniform Dial Plan feature description for more information on Uniform Dial Plan.

Hardware Requirements for Dial Plan

The Dial Plan feature requires the following hardware:

- None

Administering Dial Plan

The following steps are part of the administration process for the Dial Plan feature:

- [Displaying your dial plan](#)
- [Modifying your dial plan](#)
- [Adding extension ranges](#)
- [Adding Feature Access Codes](#)

This section describes:

- Any prerequisites for administering the Dial Plan feature
- The screens that you use to administer the Dial Plan feature
- Complete administration procedures for the Dial Plan feature

Prerequisites for administering Dial Plan

You must complete the following actions before you can administer the Dial Plan feature:

- On the **Optional Features** screen, ensure that the **Multiple Locations** field is set to **y**. If this field is not set to **y**, you cannot administer the Multi-Location Dial Plan feature. Contact your Avaya representative for assistance.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

For a complete description of the many **Optional Features** screens, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Dial Plan

Screen name	Purpose	Fields
Optional Features	Ensure that the Multi-Location Dial Plan feature is available.	Multiple Locations
Dial Plan Analysis Table	Display the dial plan.	All
	• Modify the dial plan. • Add extension ranges. • Add Feature Access Codes (FAC).	• Dialed String • Total Length • Call Type
Dial Plan Parameters	Set system-wide parameters for your dial plan.	All

Displaying your dial plan

To display your dial plan:

- 4 Type **display dialplan analysis**. Press **Enter**.

The system displays the *Dial Plan Analysis Table* screen ([Figure 157](#)).

Figure 157: Dial Plan Analysis Table screen

change dialplan analysis			DIAL PLAN ANALYSIS TABLE			Page 1 of 1		
						Percent Full: 9		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
0	1	attd	—	—	—	—	—	—
1	3	dac	—	—	—	—	—	—
3	5	ext	—	—	—	—	—	—
4	6	ext	—	—	—	—	—	—
5	7	ext	—	—	—	—	—	—
*	3	fac	—	—	—	—	—	—
#	3	fac	—	—	—	—	—	—
7	3	dac	—	—	—	—	—	—
3	4	ext	—	—	—	—	—	—
2	2	fac	—	—	—	—	—	—
—	—	—	—	—	—	—	—	—

Modifying your dial plan

In this example, you are adding a new range of dial access codes to the dial plan. You assign both FAC and TAC in the 700–799 range.

To modify your dial plan:

- 1 Type **change dialplan analysis**. Press **Enter**.

The system displays the *Dial Plan Analysis Table* screen ([Figure 157, Dial Plan Analysis Table screen](#), on page 610).

- 2 Move the cursor to an empty row.
- 3 In the **Dialed String** column, type 7. Press **Tab** to move to the next field.
- 4 In the **Total Length** column, type 3. Press **Tab** to move to the next field.
- 5 In the **Call Type** column, type **dac**.
- 6 Press **Enter** to save your changes.

Adding extension ranges

Before you assign a telephone to an extension, the extension must belong to a range that is defined in the dial plan. In this example, you are adding 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**. Press **Enter**.
The system displays the *Dial Plan Analysis Table* screen [Figure 157, Dial Plan Analysis Table screen](#), on page 610.
- 2 Move the cursor to an empty row.
- 3 In the **Dialed String** column, type **3**. Press **Tab** to move to the next field.
- 4 In the **Total Length** column, type **4**. Press **Tab** to move to the next field.
- 5 In the **Call Type** column, type **ext**.
- 6 Press **Enter** to save your changes.

Adding Feature Access Codes

A feature access code (FAC) that you assign on the *Feature Access Code (FAC)* screen must conform to your dial plan. In this example, if you want to assign a feature access code of 23 to **Last Number Dialed**, you must first add a new FAC range to the dial plan.

To add a FAC range from 20–29:

- 1 Type **change dialplan analysis**. Press **Enter**.
The system displays the *Dial Plan Analysis Table* screen [Figure 157, Dial Plan Analysis Table screen](#), on page 610.
- 2 Move the cursor to an empty row.
- 3 In the **Dialed String** column, type **2**. Press **Tab** to move to the next field.
- 4 In the **Total Length** column, type **2**. Press **Tab** to move to the next field.
- 5 In the **Call Type** column, type **fac**.
- 6 Press **Enter** to save your changes.

Reports for Dial Plan

The following reports provide information about the Dial Plan feature.

- None

Considerations for Dial Plan

This section provides information about how the Dial Plan feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Dial Plan feature under all conditions:

- You cannot assign prefixed extensions longer than five digits (including prefix) to intercom lists.
- A 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 Communication Manager automatically sets up, for example, the 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 AAR/ARS faxes. These extensions work only within Communication Manager; they do not work as remote 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 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 for Dial Plan

This section provides information about how the Dial Plan feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of the Dial Plan feature in any feature configuration.

- 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.
- Integrated Services Digital Network-Basic Rate Interface (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 Communication Manager completes the operation.
- Multifrequency (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 DCS environment:

- Extensions that differ in length from the UDP do not distribute to other media servers or 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.

The following are the interactions for Multi-location Dial Plan.

- Attendant

This feature provides a way to administer multiple attendant codes. This does not provide a way to administer multiple attendant groups. Only one attendant group is allowed per switch, unless you have attendant partitioning. This feature does provide a way to support multiple local centralized answering points. The LCAPs do not utilize attendant groups to support this.

- Attendant Vectoring

If attendant vectoring is enabled, it will take precedence over any local attendant codes administered. This is how attendant vectoring currently works. Attendant seeking or “dial 0” calls (local attendant codes, attendant code on the dial plan screen or attendant access code on the **Feature Access Code (FAC)** screen) are processed using call vectors and not using the normal attendant console call routing.

- Automatic Circuit Assurance

The field **ACA Referral Destination** on page 1 of the **System Features Parameters** screen requires either the **attd** administration on the **Dial Plan Analysis Table** screen, or administration of the attendant access code on the **Feature Access Code (FAC)** screen. This field requires that an attendant group exists.

- Automatic Wakeup

If a station has automatic wakeup requests pending when the **change extension-station** command is run, the wakeup requests are canceled.

- Call Forwarding

Any call forwarding information stored with an extension is lost when using the **change extension-station** command. If the station that is changed with the **change extension-station** command is a forwarded-to station, the extension using that forwarded-to extension must be manually updated. If not, calls are not forward correctly.

- Call Park

This feature does not support common shared extensions on the console parameter to park calls. Since common shared extensions are not assigned to physical stations, the range of common shared extensions should be able to be shared in all locations.

- Crisis Alert

The **Originating Extension** on the **Crisis Alert System Parameters** screen is not updated by the **change extension-station** command. It must be manually updated if the originating extension field was changed.

- Leave Word Calling (LWC)

The field **Stations with System-wide Retrieval Permission for the Leave Word Calling Parameters** on page 2 of the *System Features Parameters* screen require either the **attd** administration on The *Dial Plan Analysis Table* screen, or administration of the **Attendant Access Code** on the *Feature Access Code (FAC)* screen. This field requires that an attendant group exists.

- LSP

If the **change extension-station** command is run on the controller, but the translations are not also saved to the Local Spare Processor (LSP), the two system translations might not be synchronized.

- Night Service

Location-based Night Service is not supported. A customer may want to restrict attendant-seeking calls to attendants who are local to the calling party. The reason is the local attendant would most likely speak the same language as the caller. Those customers would want an attendant to be able to put one location only into Night Service, without putting the entire switch into Night Service.

One way to accomplish this is to use hunt groups as attendant queues. Each hunt group can be put into Night Service separately, and have its own Night Service destination. In addition, the Night Service destination can be administered by tenant, by trunk group, and by trunk group number.

- Station Hunting

Changing a station extension with the **change extension-station** command maintains the station hunting chain.

- Survivable Remote EPN/WAN Spare Processor

If an administrator runs the **change extension-station** command in a configuration where a survivable remote processor exists, and the command is not run on the survivable processor, when the remote processor takes control, the extensions that were changed on the server will not exist on the survivable remote.

- Uniform dial plan (UDP)

If an administrator runs the **change extension-station** command, extensions that were typed into the **UDP** screen are not updated. The change to the UDP table is handled by external system management tools that Avaya supports, but it is not changed by this command.

Distinctive Ringing

Use the Distinctive Ringing feature to distinguish between incoming call types based on the ringing pattern of the call.

Detailed description of Distinctive Ringing

This section provides a detailed description of the Distinctive Ringing feature.

You can administer the system-wide distinctive-ringing cycles for the three basic call types for users of multiappearance telephones. You cannot administer system-wide distinctive-ringing cycles for users of single-line analog telephones. For users of single-line analog telephones, you must administer the ringing cycle for the user on a **Station** screen. For more information, see the “Personalized Ringing” feature.

Most installations use a one-burst ring for internal calls, a two-burst ring for external calls, and a three-burst ring for priority calls.

There are some ringing cycles that you cannot administer. The system controls the ringing cycles for the following calls:

- Automatic and Dial Intercom
- Manual Signaling
- Redirect Notification

If an internal telephone user transfers an external call, the call usually rings as an internal call. You can administer the system so the transferred call rings as an external call.

Hardware requirements for Distinctive Ringing

The Distinctive Ringing feature requires the following hardware:

- None

Administering Distinctive Ringing

The following steps are part of the administration process for the Distinctive Ringing feature:

- [Defining Distinctive Ringing](#)

This section describes:

- Any prerequisites for administering the Distinctive Ringing feature
- The screens that you use to administer the Distinctive Ringing feature
- Complete administration procedures for the Distinctive Ringing feature

Prerequisites for administering Distinctive Ringing

You must complete the following actions before you can administer the Distinctive Ringing feature:

- None

Screens for administering Distinctive Ringing

Screen name	Purpose	Fields
<i>Feature-Related System Parameters</i>	Assign the number of rings for a priority call.	Distinctive Audible Alerting - internal, external
	Change the ringing pattern for an internal call to the ringing pattern of an external call, when a user or an attendant transfers the internal call.	Update Transferred Ring Pattern?

Defining Distinctive Ringing

To define Distinctive Ringing:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the *Feature-Related System Parameters* screen ([Figure 158, Feature-Related System Parameters screen](#), on page 617) and ([Figure 159, Feature-Related System Parameters screen](#), on page 617).

Figure 158: Feature-Related System Parameters screen

display system-parameters features	Page 5 of 14
FEATURE-RELATED SYSTEM PARAMETERS	
Public Network Trunks on Conference Call: 5	Auto Start? n
Conference Parties with Public Network Trunks: 6	Auto Hold? y
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): 0	Mode Code Interface? y
Reset Shift Timer (seconds): 0	
Station Call Transfer Recall Timer (seconds): 0	
DID Busy Treatment: tone	
Invalid Number Dialed Intercept Treatment: tone	
Allow AAR/ARS Access from DID/DIOD? n	
Allow ANI Restriction on AAR/ARS? n	
Use Trunk COR for Outgoing Trunk Disconnect? n	
7405ND Numeric Terminal Display? n	7434ND? y
DISTINCTIVE AUDIBLE ALERTING	
Internal: 1 External: 2 Priority: 3	
Attendant Originated Calls: external	
DTMF Tone Feedback Signal to VRU - Connection:	Disconnection:

Figure 159: Feature-Related System Parameters screen

change system-parameters features	page 9
FEATURE-RELATED SYSTEM PARAMETERS	
Pull Transfer: n	Update Transferred Ring Pattern? n
Outputpulse 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	
ITALIAN DCS PROTOCOL	
Italian Protocol Enabled? y	
Apply Intercept Locally? _	Enforce PNT-to-PNT Restrictions? _

- 2 Page through the screens until you see the **Distinctive Audible Alerting** area.
- 3 In the **Distinctive Audible Alerting** area, perform the following actions:
 - Type **internal** next to the number of rings that you want the system to use for an internal call.
 - Type **external** next to the number of rings that you want the system to use for an external call.

- 4 Page through the screens until you see the **Update Transferred Ring Pattern?** field.
- 5 In the **Update Transferred Ring Pattern?** field, perform one of the following actions:
 - Type **y** if you want the system to change the ringing pattern for an internal call to the ringing pattern of an external call, when a user or an attendant transfers the call.

If most of your calls go through an attendant, you might want to set this field to **y**, so that your users can identify an external call.
 - Type **n** if you do not want the system to change the ringing pattern for an internal call to the ringing pattern of an external call, when a user or an attendant transfers the call.
- 6 Press **Enter** to save your changes.

Reports for Distinctive Ringing

The following reports provide information about the Distinctive Ringing feature:

- None

Considerations for Distinctive Ringing

This section provides information about how the Distinctive Ringing feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Distinctive Ringing under all conditions. The following considerations apply to Distinctive Ringing:

- If Distinctive Ringing is disabled, the system generates a one-burst repetitive tone for all incoming calls. The one-burst repetitive tone is useful for equipment that is interfaced by analog lines, especially if you use an 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 for Distinctive Ringing

This section provides information about how the Distinctive Ringing feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Distinctive Ringing in any feature configuration.

- Personalized Ringing

The called party hears the user-selected ringing pattern for the distinctive ring cycles.

Enhanced 911

Use the Enhanced 911 (E911) feature to quickly access your local public safety agency. The public safety agency can dispatch the appropriate response team in cases of a:

- Fire
- Accident
- Crime
- Medical emergency

Enhanced 911 supports the following capabilities:

- [Configurations with media gateways in different locations](#)
- [Location Specific Routing](#)
- [E911 for wired IP telephones](#)
- [Crisis Alert](#)

Detailed description of Enhanced 911

This section provides a detailed description of the Enhanced 911 (E911) feature.

A caller who needs emergency assistance dials a Universal Emergency Number (UEN):

- 911 in the United States
- 000 in Australia
- 112 in the European community

The system routes the call through a local central office (CO), through an emergency tandem office, to the appropriate public safety answer point (PSAP). The PSAP answers the call.

A tandem office can route the call to a PSAP within four surrounding areas. In the U.S., a tandem office can route the call to four nearby area codes. If the PSAP that receives the call is not the correct PSAP to handle the emergency, the PSAP transfers the call to the correct PSAP. Transfers can only occur between geographically adjacent or nearby PSAPs.

Each PSAP usually covers one city, or one rural county or community. At the PSAP, emergency operators determine the nature of the emergency, and contact the appropriate response agency. In the U.S., a PSAP is usually responsible for an area that covers several independent police and fire departments.

With E911, the system sends the call and the Calling Party Number (CPN) over Centralized Automatic Message Accounting (CAMA) trunks. The system can also send the call and the CPN through the calling number information element (IE) over Integrated Services Digital Network (ISDN) trunks.

To learn how CAMA and ISDN trunks translate an extension to the PSAP, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

The public emergency system maintains a database that stores location and background information to help public safety agencies respond quickly with the appropriate assistance. The PSAP uses the CPN or the Caller Emergency Service Identification (CESID) number to look up the street address of the caller. The PSAP uses an Automatic Location Information (ALI) database. The ALI database is usually owned and managed by local exchange carriers (LEC). Instead of a LEC, customers can also contract with a third party to update the ALI database for them.

The E911 feature does not provide PSAP with the location of the person who placed the emergency call if the call came from a telephone that is on:

- A system that is not equipped with CAMA trunks
- An adjunct computer system that is associated with CAMA trunks

Instead, the E911 system identifies only the location of the trunk termination.

To solve this problem, you can report the emergency location extension as the CPN. After someone moves a telephone, you can manually correlate the CPN with the new telephone location. You do not have to update the ALI database for the public switched telephone network (PSTN) after each telephone move.

The E911 feature transmits the extension of a DID telephone that is associated with the calling party. The E911 feature transmits either:

- CESID over CAMA trunks
- CPN over ISDN trunks

The calling party might be at or near a telephone on a remote port network. The calling party might also be at a remote location that is served by an off-premises telephone.

For Avaya IP Softphone, if the **Remote Softphone Emergency Calls** field on the **Station** screen is set to **extension**, the system uses the entry in the **Emergency Location Ext** field as the E911 CPN instead of the extension of the IP Softphone. For more information on Avaya IP Softphone, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Configurations with media gateways in different locations

Media gateways in different locations have a different PSAP than the PSAP of the main server. You must ensure that the system correctly routes emergency calls to the correct PSAP that are made from telephones that are registered to each gateway.

To ensure that a media gateway in a separate location can route emergency calls properly, the system requires a CO trunk or a Primary Rate Interface (PRI) trunk from the gateway to the LEC. You must administer each media gateway that is in a different PSAP jurisdiction than the main server in a separate location. This separate administration ensures that the system can route emergency calls from that location. If a media gateway is in the same PSAP jurisdiction as the main server, you do not need to administer the media gateway in a separate location.

Location Specific Routing

In a configuration with one location, the system routes all outgoing calls to the PSTN according to the Automatic Route Selection (ARS) table for location 1. If media gateways are in different locations, then you must administer Location Specific Routing for each location.

If a media gateway is in a separate location from the main server, you can administer the media gateway:

- In locations 2-64 for Linux servers
- In locations 2-44 for Avaya DEFINITY servers

The command for Location Specific Routing is **change ars analysis location X 0**, where **X** is the chosen location (2-64 or 2-44), and **0** is the first placeholder in the analysis table. Once the system displays the table, enter the routing information:

- Chosen dialed string (probably 911)
- Min and max digits
- Route pattern
- Call type

You must also set up the required route pattern information. When you complete this administration, emergency calls from the location route over a CO or PRI trunk to the LEC.

- 1** If a media gateway is connected to the LEC over a CO trunk, the extension of the CO trunk identifies the gateway service address. The service address is the physical location that the PSAP sees. Calls from any telephone that is registered to the gateway display the CO trunk extension to the PSAP. The PSAP then sends the emergency response to the service address.

Use the CO trunk to send and receive emergency calls only. You must verify with the LEC that the extension of the CO trunk is in the PSAP database so that all emergency help is sent to the correct location.
- 2** If a media gateway is connected to the LEC over a PRI trunk, each extension that is registered to the gateway sends a separate extension to the PSAP. You must verify with the LEC that all the extensions are in the PSAP database.
- 3** If you use DID numbers from the main server to administer the extensions at the gateway, the numbers that are used at the media gateway must be moved to the CO where the 911 calls terminate. If the numbers are not moved, emergency help is sent to the main server location and not to the media gateway location.

If the main server and the media gateway are in different LEC jurisdictions:

- If you use a CO trunk from the gateway to the central office, follow the procedure described in [Step 1](#) above. Ensure that the correct extension is in the PSAP database for the CO trunk. All calls that are made from the gateway use this extension.
- If you use a PRI trunk, you must purchase a block of DID extensions from the LEC for your gateway telephones. The gateway extensions can be shortened for private calls within your system. For example, you can shorten extension 765-4321 to 4321 for private calls within your system.

When the system calls the PSTN, you must use the complete extension so that the number is recognized at the PSAP. For example, the system must send extension 765-4321 to the PSAP. The PSAP does not recognize extension 4321.

These two scenarios apply to each media gateway that is in a separate location. You must repeat each applicable procedure for each media gateway that is connected to a server with a unique PSAP.

You can use the same Location Specific Routing tables and trunks for:

- Media gateways that are in the same location as the main system
- More than one media gateways that are in the same location

Contact your Avaya representative if you have questions about these procedures or scenarios.

E911 for wired IP telephones



CAUTION:

When someone dials an emergency number from an Avaya IP telephone, the emergency call reaches the local emergency service in the PSAP only when the telephone system has local trunks. If someone dials an emergency number from a remote location that does not have local trunks, an Avaya IP telephone cannot dial to and connect with local emergency services.

To avoid possible delays in getting emergency aid, *do not use an Avaya IP telephone to dial emergency numbers from a remote location*. Avaya is not responsible or liable for any damages that might result from misplaced emergency calls that someone makes from an Avaya IP telephone.

Your use of this software indicates that you have read this advisory. You further agree to use an alternative telephone to dial all emergency calls from remote locations. Contact your Avaya representative if you have questions about emergency calls from an IP telephone.

When someone dials an emergency number from a wired IP telephone, the system assigns an Emergency Location Information Number (ELIN) through an IP subnetwork, or subnet. The system then sends the ELIN over either CAMA or ISDN PRI trunks to the emergency services network. To use this capability, you must have subnets that correspond to geographical areas.

The E911 for wired IP telephones capability works with two types of IP protocols:

- H.323
- SIP

If someone dials an emergency number from a wired IP telephone, the system at the PSAP uses the CPN to look up the physical location of the caller. However, the CPN might not always correspond to the physical location of the caller, because users with:

- H.323 IP telephones can move the telephones without notifying the system administrator
- SIP IP telephones can use the same extension simultaneously at several different telephones

Without the E911 for wired IP telephones capability, the emergency response personnel might go to the wrong physical location. With the E911 for wired IP telephones capability, the system properly identifies the location of the caller. The emergency response personnel can now go to the correct physical location, even if an emergency call comes from a bridged call appearance.

Whenever someone uses an IP telephone to dial an emergency number, the software compares the following two values for that IP telephone:

- The **Emergency Location Extension** field on the **IP Address Mapping** screen
- The **Emergency Location Ext** field on the **Station** screen
 - If the two values are the same, the telephone most likely has not moved. If the telephone has moved, the telephone has moved within the same subnet.
 - If the two values are different, and if the **Emergency Location Extension** on the **IP Address Mapping** screen is *not* blank, the telephone has moved from one subnet to another.
 - If the **Emergency Location Extension** on the **IP Address Mapping** screen is blank, then the administrator expects the caller to be located outside the LAN. This situation is true for a softphone.

Whenever you add an extension as an **Emergency Location Extension** to the **IP Address Mapping** screen, check all of the **Station** screens for telephones in that IP address range.

- If the telephone is a DID number, make sure that the **Station** screen has the same **Emergency Location Extension** as does the **IP Address Mapping** screen.
- If the telephone is not a DID number, make sure that the **Station** screen has a different **Emergency Location Extension** as does the **IP Address Mapping** screen.

Emergency Extension Forwarding

If an emergency call should drop (get disconnected), the public safety personnel will attempt to call back. If the ELIN that was sent was not equivalent to the caller's extension number, the return call would ring some other set than the one that dialed 911. To overcome that limitation, you can automatically forward that return call to the set that placed the emergency call for an administered period of time.

The **Emergency Extension Forwarding (min)** field on the **Feature-Related System Parameters** screen allows you to set the **Emergency Extension Forwarding** timer for all incoming trunk calls if an emergency call gets cut off (drops).

This **Emergency Extension Forwarding** only applies if the emergency location extension number is an extension on the same system as the extension that dialed 911. Customers who have several systems in a campus should assign several emergency location extensions.

Call Forwarding of dropped emergency calls

Suppose that an IP telephone and a nearby media gateway that provides a connection to the PSTN are both registered to a primary server. Suppose that the media gateway is backed up by an LSP, and that the IP telephone has some LSPs in its alternate gatekeeper list. Suppose that the IP telephone dials 911, and then the LAN crashes and partially recovers. If both the IP telephone and the media gateway can re-register to the primary server, everything is fine. If both the IP telephone and the media gateway are forced to register with the same LSP, everything is fine.

But if one of them re-registers to one LSP and the other re-registers to the primary server, or if they re-register to different LSPs, there's a minor problem. When the emergency response personnel call back, the server that the media gateway is registered with will think that the IP telephone is unregistered. The media gateway's server will not be able to forward the call to the IP telephone. Instead, the media gateway's server will attempt to ring the call at the telephone equivalent to the sent ELIN.

Even with automatic emergency return call forwarding, pick the **Emergency Location Extension** that is administered in the **ip-network-map** screen as follows:

- On media gateways having incoming PSTN trunks
- On the same media gateway as the telephones that they cover, assuming that the media gateway has incoming PSTN trunks

Crisis Alert

The Crisis Alert capability notifies the attendant, up to 10 other designated users, or a digital pager when someone dials an emergency number. Designated users might include:

- Security guards
- Receptionists
- Secretaries
- Front office personnel
- Human Resources personnel

If a person dials an emergency number, the attendant or other designated users might want to know who made the call so that they can direct the emergency personnel to the right place.

When a user dials an emergency number, the system sends both an audible alert and a visual alert to the telephone of the designated user.

- The audible alert is a siren alarm.
- The visual alert is a flashing **crss-alert** button. The system also displays the name and extension of the caller.
 - If a user, John Doe at extension 3041, dialed an emergency number, the following information appears on the display of attendant consoles that have a **crss-alert** button:
E=JOHN DOE 3041 EM
 - If a user, John Doe at extension 3041, dialed an emergency number, the following information appears on the display of digital telephones that have a **crss-alert** button:
EM=JOHN DOE 3041 EM

Cancelling an alert

To cancel the alert, you can administer the system so that only one notified user must acknowledge the alert, or all notified users must acknowledge the alert.

To cancel an emergency alert, an attendant must press the **crss-alert** button on the attendant console three times:

- The first press turns off the siren alarm.
- The second press stops the **crss-alert** lamp from flashing.
- The third press clears the display.

Digital telephone users must press the **crss-alert** button on the telephone to cancel the emergency alert.

- If only one user must acknowledge the alarm, the siren alarm stops at all telephones.
- If all administered users must acknowledge the alarm, the alarm continues at each telephone until the user of that telephone presses the **crss-alert** button. Once all administered users acknowledge the alarm, the siren alarm stops.
- The name and the extension of the person who dialed the emergency number remains on the telephone display. To completely cancel an alert and clear the display, each administered user must press the **normal** button.

If someone makes an emergency call while another crisis alert is still active, the second emergency call is placed in a queue.

- If you administer the system so that all users must acknowledge the alert, all users must acknowledge all emergency calls. The calls might not appear in the queue in the order that the calls were made.
- If you administer the system so that only one user must acknowledge the alert, the first alert remains active at the telephone from where the alert was acknowledged. Any subsequent calls are queued to the next available telephone, in the order that the calls were made.

Once you administer the Crisis Alert capability, the system continues to record each emergency call. The system also sends a record to the system printer, if a system printer is available. If a system printer is not available, type **list emergency** to view the *Emergency Access Calls* report.

Hardware requirements for Enhanced 911

The Enhanced 911 feature requires the following hardware:

- The E911 for wired IP telephones capability requires a wired IP telephone.
- The Crisis Alert capability requires:
 - An attendant console
 - An optional display telephone
 - An optional digital pager

Administering Enhanced 911

In this example, you administer your system to notify the attendant, plus the security guard at three entrances. All three guards and the attendant must acknowledge the alarm to stop the alarm.

The following steps are part of the administration process for the Enhanced 911 feature:

- [Setting up Crisis Alert to an attendant or a display telephone](#) on page 627
- [Setting up Crisis Alert to notify a digital pager](#) on page 631
- [Setting up emergency extension forwarding](#) on page 633
- [Setting up CAMA numbering](#) on page 634

This section describes:

- Any prerequisites for administering the Enhanced 911 feature
- The screens that you use to administer the Enhanced 911 feature
- Complete administration procedures for the Enhanced 911 feature

Prerequisites for administering Enhanced 911

You must complete the following actions before you can administer the Enhanced 911 feature:

- Ensure that you have set up calling party restrictions on the **Class of Restriction** screen. For information on how to set up a Class of Restriction (COR), see the “Class of Restriction” feature.
- Ensure that you have set up ARS access codes on the **Feature Access Code (FAC)** screen. For information on how to set up a Feature Access Code (FAC), see the “Feature Access Code” feature.
- Ensure that you have set up all telephone route patterns on the **Route Pattern** screen. For information on how to set up a route pattern, see the “Uniform Dial Plan” feature.

Screens for administering Enhanced 911

Screen name	Purpose	Fields
ARS Digit Analysis Table	Set up an emergency number that users dial to access emergency services.	All
Attendant Console	Notify the attendant when someone dials the emergency number.	Any button field in the Feature Button Assignments area.
CAMA Numbering - E911 Format	Format CAMA trunks for dialing.	All
Class of Restriction	Ensure that you have set up calling party restrictions on your system.	Calling Party Restriction
Crisis Alert System Parameters	Force all users who have a crss-alert button on the telephone to acknowledge a crisis alert.	Every User Responds
	Administer the Crisis Alert capability to send an alert to a digital pager.	All fields in the Alert Pager area

Screen name	Purpose	Fields
Feature Access Code (FAC)	Ensure that you have set up FACs for ARS access codes on your system.	- ARS Access Code 1 - ARS Access Code 2
Feature-Related System Parameters	Set the timer to forward all incoming trunk calls in an emergency.	Emergency Extension Forwarding (min)
Optional Features	Ensure that your system is set up to handle Automatic Route Selection (ARS) routing and digit analysis.	ARS
Route Pattern	Ensure that you have set up route patterns on your system.	All
Station	Notify another user when someone dials the emergency number.	Any button field in the Feature Button Assignments area.
	Prevent the use of an Avaya IP Softphone extension as the E911 CPN.	- Remote Softphone Emergency Calls - Emergency Location Ext

Setting up Crisis Alert to an attendant or a display telephone

To activate the Crisis Alert feature, you must complete the following procedures:

- [Setting up the emergency number](#) on page 627
- [Setting up the attendant console to receive emergency notification](#) on page 629
- [Setting up digital telephones to receive emergency notification](#) on page 629
- [Setting which users must acknowledge the emergency alert](#) on page 630

Setting up the emergency number

Prerequisites

You must complete the following actions before you can set up the emergency number:

- On the **Optional Features** screen, ensure that the ARS field is set to **y**. If this field is not set to **y**, contact your Avaya representative for assistance.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

To set up the emergency number:

- 1 Type **change ars analysis n**, where **n** is the number of the ARS table that you want to change. Press **Enter**.

The system displays the **ARS Digit Analysis Table** screen ([Figure 160, ARS Digit Analysis Table screen](#), on page 628).

Figure 160: ARS Digit Analysis Table screen

change ars analysis 5
Page 1 of 2

ARS DIGIT ANALYSIS TABLE
 Location: all Percent Full: 6

Dialed String	Total Min Max	Route Pattern	Call Type	Node Num	ANI Reqd
911	3 3	1	alrt	—	n
11	2 2	1	alrt	—	n
	— —	—	—	—	n
	— —	—	—	—	n
	— —	—	—	—	n
	— —	—	—	—	n
	— —	—	—	—	n
	— —	—	—	—	n
	— —	—	—	—	n
	— —	—	—	—	n
	— —	—	—	—	n
	— —	—	—	—	n
	— —	—	—	—	n
	— —	—	—	—	n
	— —	—	—	—	n
	— —	—	—	—	n

- 2 In the **Dialed String** field, type the number that users dial to reach emergency services.
In this example, type **911**.
- 3 In the **Total Min** and **Total Max** fields, type the number of digits that you entered in the **Dialed String** field.
In this example, type **3**. The user must dial all 3 digits in the **Dialed String** field for the system to treat the call as an emergency call.
- 4 In the **Route Pattern** field, type the number of the route pattern for local calls.
In this example, type **1**.
- 5 In the **Call Type** field, type **emer** or **alrt**.
 - **emer** identifies that the number in the **Dialed String** field as an emergency call.
 - **alrt** identifies that the number in the **Dialed String** field activates emergency alert notification.



Important:

If you use the digit **9** on the **Dialplan Analysis Table** screen as the ARS access code, also administer the dial string **11** as either an **emer** or **alrt** number. That way, when a user dials **911**, the digit **9** provides an outside line, and the digits **11** indicate an emergency or crisis alert call.

- 6 Press **Enter** to save your changes.

Setting up the attendant console to receive emergency notification

When Crisis Alerting is active at the attendant console, the console is in position-busy mode. No other incoming calls interfere with the emergency call, but the console can still originate calls. The attendant must press the **position-busy** button to unbusy the console. The attendant must then press the **crss-alert** button to deactivate the audible and the visual alerts.

To set up the attendant console to receive emergency notification:

- 1 Type **change attendant *n***, where *n* is the number of the attendant console. Press **Enter**.
The system displays the **Attendant Console** screen.
- 2 Press **Next** until you see the Feature Button Assignments area ([Figure 161, Attendant Console screen](#), on page 629).

Figure 161: Attendant Console screen

change attendant 1	Page 3 of 4
ATTENDANT CONSOLE	
FEATURE BUTTON ASSIGNMENTS	
1: split	13:
2: dont-split	14:
3:	15: whisp-act
4:	16:
5:	17:
6: hold	18:
7:	19: forced-rel
8:	20:
9:	21: crss-alert
10:	22:
11:	23: night-serv
12:	24: pos-busy

- 3 In the Feature Button Assignments area, assign **crss-alert** to a button.
In this example, we assign **crss-alert** to button 21.
- 4 Press **Enter** to save your changes.

Setting up digital telephones to receive emergency notification

Prerequisites

You must complete the following actions before you can set up digital telephones to receive emergency notification:

- On the **Station** screen for each telephone that you want to receive emergency notification:
 - Ensure that the extension is a digital display telephone
 - Check the **Type** field to ensure that the telephone is not a virtual extension

To view the **Station** screen, type **change station *n***, where *n* is the extension. Press **Enter**.

To set up the digital telephone of each security guard to receive emergency notification:

- 1 Type **change station *n***, where *n* is the extension of the security guard. Press **Enter**.
The system displays the **Station** screen.
- 2 Press **Next** until you see the Feature Button Assignments area ([Figure 162, Station screen](#), on page 630).

Figure 162: Station screen

change station 5671000	Page 3 of 5
STATION	
FEATURE BUTTON ASSIGNMENTS	
9: crss-alert	
10:	
11:	
12:	
13:	
14:	
15:	
16:	

- 3 In the Feature Button Assignments area, assign **crss-alert** to a button.
In this example, we assign **crss-alert** to button 9. You cannot assign this button to a softkey.
- 4 Press **Enter** to save your changes.
- 5 Repeat this process for the telephone of each security guard.

Setting which users must acknowledge the emergency alert

To force all security guards and the attendant to acknowledge and cancel an alert:

- 1 Type **change system-parameters crisis-alert**. Press **Enter**.
The system displays the **Crisis Alert System Parameters** screen ([Figure 163, Crisis Alert System Parameters screen](#), on page 630).

Figure 163: Crisis Alert System Parameters screen

change system-parameters crisis-alert	Page 1 of 1
CRISIS ALERT SYSTEM PARAMETERS	
ALERT STATION	
Every User Responds? y	
ALERT PAGER	
Alert Pager? n	

- 2 In the **Every User Responds** field, type **y**.
If you set the **Every User Responds** field to **n**, anyone of the designated users can cancel an alert.
- 3 Press **Enter** to save your changes.

Setting up Crisis Alert to notify a digital pager

You also have the option to have the Crisis Alert capability notify a digital pager. When someone dials an emergency number, the system sends the extension and the location of the caller to the administered pager.

This feature sends a message of 7 to 22 digits to the administered pager. This message includes a crisis alert code, and an extension or a room number. You can also administer a main number so that the pager displays the location from where the emergency call originated.

To receive a crisis alert message, you must administer at least one attendant console or one digital telephone with a **crss-alert** button.

The crisis alert call to a digital pager uses two to four trunks. The system uses:

- One trunk for the actual call
- One to three trunks to notify one to three pagers, depending on how many pagers you administer.

Prerequisites

You must complete the following actions before a user can receive notification to a digital pager in an emergency:

- In the **ARS Digit Analysis Table** screen, you must have emergency numbers in the **Call Type** column set to **alrt**. For more information, see [Setting up the emergency number](#) on page 627.
- You must administer a **crss-alert** button on at least one of the following telephones:
 - Attendant console. For more information, see [Setting up the attendant console to receive emergency notification](#) on page 629.
 - Digital telephone. For more information, see [Setting up digital telephones to receive emergency notification](#) on page 629.
- You must also have at least one digital numeric pager, and one of the following circuit packs:
 - Call Classifier
 - Tone-Clock with Call Classification and Tone Detection

To set up Crisis Alert to notify a digital pager:

- 1 Type **change system-parameters crisis-alert**. Press **Enter**.
The system displays the **Crisis Alert System Parameters** screen ([Figure 164, Crisis Alert System Parameters screen](#), on page 632).

Figure 164: Crisis Alert System Parameters screen

change system-parameters crisis-alert	Page 1 of 1
CRISIS ALERT SYSTEM PARAMETERS	
ALERT STATION	
Every User Responds? y	
ALERT PAGER	
Alert Pager? y	
Originating Extension: 7768	
Crisis Alert Code: 911	
Retries: 5	
Retry Interval (sec): 15	
Main Number: 3035550800	
Pager Number	Pin Number
1: 3035559001	1: pp77614567890
2:	2:
3:	3:
DTMF Duration - Tone (msec): 100 Pause (msec): 100	

- 2 In the Alert Pager field, type **y**.
The system displays additional Crisis Alert administration fields.
- 3 In the Originating Extension field, type a valid unused extension to send the Crisis Alert message.
In this example, type **7768**.
- 4 In the Crisis Alert Code field, type the number that a user dials to call for emergency assistance.
In this example, type **911**.
- 5 In the Retries field, type the number of additional times that you want the system to try to send the alert message in case of an unsuccessful first attempt.
In this example, type **5**.
- 6 In the Retry Interval (sec) field, type the number of seconds between retries.
In this example, type **15**.
- 7 In the Main Number field, type the number that you want the system to display at the end of the pager message.
In this example, type **3035550800**.
- 8 In the Pager Number field, type the telephone number for the pager.
In this example, type **3035559001**.
- 9 In the Pin Number field, type the personal identification number (PIN), if required, for the pager. Insert pause digits (**pp**) as needed to pause for announcements from the pager service to complete before sending the PIN.
In this example, type **pp77614567890**.
- 10 In the DTMF Duration - Tone (msec) field, type the the number of milliseconds that the DTMF tone plays for each digit.
In this example, type **100**.

- 11** In the **Pause (msec)** field, type the number of milliseconds between DTMF tones for each digit.
In this example, type **100**.
- 12** Press **Enter** to save your changes.

Setting up emergency extension forwarding

If an emergency call gets disconnected, the public safety person that you were talking to attempts to call you back. If the ELIN that the PSAP receives is not equivalent to the extension of the caller, the return call might ring at a different telephone. To solve this problem, you can automatically forward all incoming trunk calls, for an administered period of time, to the telephone that placed the emergency call.

You must forward all incoming trunk calls because the system cannot determine what incoming trunk call is from the PSAP.

To set up emergency extension forwarding:

- 1** Type **change system-parameters features**. Press **Enter**.
The system displays the *Feature-Related System Parameters* screen.
- 2** Press **Next** until you see the **Emergency Extension Forwarding (min)** field
([Figure 165, Feature-Related System Parameters screen](#), on page 633).

Figure 165: Feature-Related System Parameters screen

change system-parameters features	Page 5 of 14
FEATURE-RELATED SYSTEM PARAMETERS	
SYSTEM PARAMETER PRINTERS	
System Printer Endpoint:	Lines Per Page: 60
Emergency Extension Forwarding (min): 10	
SYSTEM-WIDE PARAMETERS	
Switch Name: Norbert	
MALICIOUS CALL TRACE PARAMETERS	
Apply MCT Warning Tone? y	MCT Voice Recorder Trunk Group:
Delay Sending RElease (seconds)? 0	
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:

- 3** In the **Emergency Extension Forwarding (min)** field, type the number of minutes that you want the system to forward all incoming trunk calls to the emergency extension.

The timer starts once an emergency call gets disconnected. This timer applies only if the emergency location extension is an extension on the same system as the extension that dialed 911. Customers with several systems in a location must assign multiple emergency location extensions.
- 4** Press **Enter** to save your changes.

Setting up CAMA numbering

Use the **CAMA Numbering - E911 Format** screen to administer Centralized Automatic Message Accounting (CAMA) trunks. Also use this screen to provide Caller Emergency Service Identification (CESID) information to your local emergency system through the local tandem office.

This screen provides the CESID format by extension or number blocks. This flexibility allows for multiple CESID formats to be sent over multiple CAMA trunk groups, and mixed telephone numbering plans. This flexibility also allows some limited conversion from non-DID to DID numbers that the Private Switch/Automatic Location Interface (PS/ALI) database usually requires.

The **System CESID Default** field defines the CESID for all extensions that are not defined in the **Ext Code** fields. The first page of this screen contains the default CESID, plus extension fields for CESID entries. Each remaining page contains additional extension fields for CESID entries.

NOTE:

The following procedure assumes that you already set up a CAMA trunk group. For information on how to set up trunk groups, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

To set up CAMA numbering:

- 1 Type **change cama-numbering**. Press **Enter**.

The system displays the **CAMA Numbering - E911 Format** screen ([Figure 166, CAMA Numbering - E911 Format screen](#), on page 634).

Figure 166: CAMA Numbering - E911 Format screen

change cama-numbering				CAMA NUMBERING - E911 FORMAT				Page 1 of 15	
System CESID Default: _____									
Ext Len	Ext Code	CESID	Total Length	Ext Len	Ext Code	CESID	Total Length		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		
-	_____	_____	___	-	_____	_____	___		

- 2 In the **System CESID Default** field, type the CESID that the system sends over the CAMA trunk if you do not define the **Ext Code** fields.

Enter a number of 1 to 16 digits. The default value is blank.

- 3 In the **Ext Len** field, type the number of digits in the extension.
Valid entries are from 1 to 5. The default value is blank.
- 4 In the **Ext Code** field, type the leading digits or all of the digits in the extension for the specified CESID.

If the value in the **Ext Len** field is greater than the number of digits in the **Ext Code** field, the system interprets the **Ext Code** as a block of digits. For example, if the value in the **Ext Len** field is 4 and the value in the **Ext Code** field is 11, the CESID serves extensions 1100 through 1199. The **Ext Code** [11] is for a DID block. **Ext Code** [126] might point a non-DID block to a nearby DID extension 5241666.

Enter a number of 1 to 5 digits. The default value is blank.
- 5 In the **CESID** field, type the number that identifies the calling terminal within an emergency service system. This field can represent a prefix to an extension, or the entire CESID.

Enter a number of 1 to 16 digits. The default value is blank.
- 6 In the **Total Length** field, type the total number of digits to send.

Enter a number of 1 to 16 digits. The default value is blank.
- 7 Press **Enter** to save your changes.

Reports for Enhanced 911

The following reports provide information about the Enhanced 911 feature:

- The **Emergency Access Calls** report shows the following information for each emergency call:
 - Extension
 - Event
 - Type of call
 - Time

For detailed information on this report and the associated commands, [click here](#), or see *Reports for Avaya Communication Manager*.

Considerations for Enhanced 911

This section provides information about how the Enhanced 911 feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Enhanced 911 under all conditions.

The following considerations apply to Enhanced 911:

- The Enhanced E911 feature only applies to emergency calls that go over CAMA and ISDN trunks.
- Provide several call appearances on the last telephone in both the coverage path and the telephone hunting path of the Emergency Location Extension.
- Do not include voice mail, automated attendant, or announcement extensions for Emergency Location Extensions.

The following two consideration scenarios apply to a local spare processor (LSP), and to backup S8700 servers. These considerations apply if the spare processor is an asynchronous transfer mode wide area network (ATM WAN), or is on a survivable remote expansion port network (EPN).

- Sending the correct ELIN to the PSAP

Once each day, the LSP copies administration translations from its primary server. The system never copies translations from the LSP to the primary server. IP telephones can stay under the control of an LSP for either 6 days or 10 days, or forever. The length of time depends on what version of software the customer is running.

If someone notifies you that an IP extension moved, update the ALI database with the physical location of the IP extension. Also change the **Emergency Location Extension** field on the **IP Address Mapping** screen to match the new subnet of the IP extension.

If the telephone registers with the LSP before the system copies the translations from the main server, the **Emergency Location Extension** field is still set to the old value. If the user dials 911, emergency response personnel might go to the approximate location of the caller instead of to the exact location.

Never update translations directly on the LSP.

The following considerations apply to the Crisis Alert capability:

- The Automatic Number Identification (ANI) that the system sends to the CO might not be the same extension as the telephone that the person used to dial the emergency. If the call is disconnected and the emergency personnel call back, the emergency person calls the ANI. The emergency person might not reach the caller.

If a telephone has a **crss-alert** button assigned, the return call is answered by someone who was notified of the extension that made the emergency call. That person can then forward the return call from the emergency person to the extension that made the emergency call.

- Only one **crss-alert** button can appear on an attendant console or on a digital telephone.
- Attendant consoles or digital telephones without a **crss-alert** button do not receive emergency notification.

Interactions for Enhanced 911

This section provides information about how the Enhanced E911 feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of E911 in any feature configuration.

- Bridged Call Appearances

Emergency 911 calls from a bridged extension report the CESID of the principle telephone.

- Class of Restriction (COR), Tenant Partitioning, and Facilities Restriction Level (FRL)

COR, Tenant Partitioning, and FRL restrictions apply to CAMA trunks and 911 calls.

- Expert Agent Selection (EAS)

When an EAS agent dials 911, the CESID is the number of the physical telephone, not of the logical agent.

- Personal Station Access/Terminal Translation Initialization (PAS/TTI)

When a PAS/TTI telephone is associated with a physical telephone, the CESID is that of the PSA/TTI telephone, not of the physical telephone.

- Network Address Translation (NAT) devices

If a telephone is operating behind a NAT device, the ELIN is the translated IP address of the telephone, and not the native IP address. If a customer is using a NAT device, the NAT devices have to preserve IP subnets. To require a change in IP address, the E911 for IP Wired Extensions feature expects a move from one geographic area to another.

For example, a customer has 2 subnets, 1.1.1.* and 2.2.2.*, that are mapped to two different ELINs in the *ip-network-map* screen. Both of these subnets are operating behind a NAT device. If the NAT device maps both subnets of 1.1.1.* and 2.2.2.* into addresses in the range 3.3.3.*, the software can never detect if someone moves a telephone from 1.1.1.* to 2.2.2.*.

- Non-ARS calls

The E911 feature uses ARS digit analysis to classify an outgoing call as an emergency or crisis alert call. If a user dials an emergency call without using ARS — for example, using a trunk access code, personal CO line button, facility test call, or AAR access code if AAR digit analysis does not overflow to ARS — this feature does not apply.

The following interactions apply to the Crisis Alert capability:

- Centralized Attendant Service (CAS)

If CAS is enabled, the emergency alert still goes to the local attendant.

- Outgoing Trunk Queuing

If a user attempts to make an emergency call when all trunks are busy, the call does not generate an alert. If the Outgoing Trunk Queuing feature is enabled for a trunk group, the call is placed in queue, but does not generate an alert.

- Tenant Partitioning

If Tenant Partitioning is active, attendants receive emergency notification from only those callers who are within the tenant partition. If no attendant is assigned to a partition from which an emergency call originates, the system still sends a record of the call to the system printer, and to the *Emergency Access Calls* report.

- Terminal Self-Administration

Users who can administer their own telephones cannot disable a **crss-alert** button.

Extended User Administration of Redirected Calls

Use Extended User Administration of Redirected Calls to change your lead-coverage path or your call forwarding extension from any on-site or off-site telephone.

Detailed description of Extended User Administration of Redirected Calls

With Extended User Administration of Redirected Calls, a user can change the lead coverage path or the call forwarding extension from any on-site or off-site telephone.

The Extended User Administration of Redirected Calls feature does not change the Call Coverage feature, the Call Forwarding All Calls, or Call Forwarding Busy/Don't Answer. Extended User Administration of Redirected Calls merely allows a user to select between one of two previously administered coverage paths, or to change the forwarding extension from any telephone.

A user must enter both a Feature Access Code (FAC) and a Station Security Code (SCC) to use Extended User Administration of Redirected Calls from an:

- On-site telephone that is not assigned to the user
- Off-site telephone



SECURITY ALERT:

The system logs invalid extensions and invalid station security codes (SCCs) as security violations. If you enable Security Violation Notification, the system displays the following information on the **Monitor Security-Violations Station Security Codes** screen or report:

- The extension or the incoming trunk from which the user dialed the command sequence
- The Feature Access Code (FAC)
- The command string that the user dialed

See the “Security Violation Notification” feature for more information.

Disabling the telecommuting access extension

If you want to quickly disable Extended User Administration of Redirected Calls for all users, change the Telecommuting Access Extension field on the **Telecommuting Access** screen to blank.

Extended User Administration of Redirected Calls and DCS

If your users operate in a Distributed Communication System (DCS) environment, you must assign a different telecommuting access extension to each server. You must tell your users which extension to use for telecommuting access.

A user can use Extended User Administration of Redirected Calls from any of the DCS nodes. A user must dial the telecommuting access extension of the node on which the user telephone is defined, before the user can use any of the extended FACs.

Extended User Administration of Redirected Calls and Class of Service

The following table shows the relationship between Class of Service (COS) and the ability to forward calls from the telephone that is assigned to a user without a security code, or from any on-site or off-site telephone with a security code.

Table 55: COS and Extended User Administration of Redirected Calls and Call Forwarding

If the COS of the user are set as follows:				The user can:	
Call Fwd All COS	Call Fwd B/DA COS	Extended Call Fwd Activate All COS	Extended Call Fwd Activate Busy D/A COS	Forward calls from the user telephone without a security code	Forward calls from the user telephone or from another telephone with a security code
Yes	Yes	Yes	Yes	Yes	Yes
No	No	Yes	Yes	No	Yes
Yes	Yes	No	No	Yes	No

Extended User Administration of Redirected Calls and COR

The user Class of Restriction (COR) controls the use of the change coverage option of Extended User Administration of Redirected Calls. If the **Can Change Coverage** field on the **COR** screen is set to **y**, the user can use the FAC for Change Coverage Feature Access Code to change the coverage option.

Extended User Administration of Redirected Calls from an off-site telephone

To use Extended User Administration of Redirected Calls from an off-site telephone, a user must first access the telecommuting access extension. If the user makes the request through Direct Inward Dialing (DID), the user must precede the extension with the correct public network prefix. If the user makes the request through a trunk group that is dedicated to remote access, the user must dial the public network number for the trunk group.

The system provides dial tone after the user accesses the telecommuting access extension. After the dial tone, the user can enter only one of the four following FACs that are associated with Extended User Administration of Redirected Calls:

- Extended Call Fwd All Activate
- Extended Call Fwd Busy D/A Activate
- Extended Call Fwd Deactivation
- Change Coverage

Hardware requirements for Extended User Administration of Redirected Calls

The Extended User Administration of Redirected Calls feature requires the following hardware:

- None

Administering Extended User Administration of Redirected Calls

The following steps are part of the administration process for the Extended User Administration of Redirected Calls feature requires the following hardware:

- [Assigning a station security code to a user](#)

This section describes:

- Any prerequisites for administering the Extended User Administration of Redirected Calls feature
- The screens that you use to administer the Extended User Administration of Redirected Calls feature
- Complete administration procedures for the Extended User Administration of Redirected Calls feature

Prerequisites for administering Extended User Administration of Redirected Calls

You must complete the following actions before you can administer the Extended User Administration of Redirected Calls feature:

- Ensure that the feature is enabled for your system.
- Assign a telecommuting access extension for your system.
- Assign the extended Feature Access Codes (FACs) for your system.
 - Extended Call FWD Activates Busy D/A
 - Extended Call Fwd Activate All
 - Extended Call Fwd Deactivation
 - Change Coverage
- Assign the Class of Service (COS) for extended forwarding.
- Assign a Class of Restriction (COR) that allows a user to change coverage from an on-site or an off-site telephone.
- View the **Optional Features** screen, and ensure that the Cvg Of Calls Redirected Off-Net field is set to **y**. If the Cvg Of Calls Redirected Off-net field is set to **n**, your system is not enabled for the Extended User Administration of Redirected Calls feature. Contact your Avaya representative before you continue with this procedure.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

To assign a telecommuting access extension for your system:

- 1 Type **add telecommuting-access**. Press **Enter**.

The system displays the **Telecommuting Access** screen ([Figure 167, Telecommuting Access screen](#), on page 642).

Figure 167: Telecommuting Access screen

<pre>add telecommuting-access TELECOMMUTING ACCESS Telecommuting Access Extension: _____</pre>

- 2 In the Telecommuting Access Extension field, type a one-digit to seven-digit extension that conforms to the your system dial plan.
- 3 Press **Enter** to save your changes.

To assign the extended FACs for your system:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the **Feature Access Codes (FAC)** screen ([Figure 168, Feature Access Code \(FAC\) screen](#), on page 643) and ([Figure 168, Feature Access Code \(FAC\) screen](#), on page 643).

Figure 168: Feature Access Code (FAC) screen

change feature-access-codes	FEATURE ACCESS CODE (FAC)	Page 1 of 7
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: _____ Attendant 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: _____		

Figure 169: Feature Access Code (FAC) screen

display feature-access-codes	FEATURE ACCESS CODE (FAC)	Page 2 of 7
Data Origination Access Code: _____ Data Privacy Access Code: _____ Directed Call Pickup Access Code: _____ Emergency Access to Attendant Access Code: _____ Enhanced EC500 Activation: *45 Deactivation: *46 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: *03 Leave Word Calling Message Retrieval Lock: *33 Leave Word Calling Message Retrieval Unlock: *34 Leave Word Calling Send A Message: *35 Leave Word Calling Cancel A Message: *37 Malicious Call Trace Activation: Deactivation: _____ Meet-me Conference Access Code Change: _____		

- 2 In the Change Coverage Access Code field, type the FAC to change a coverage path from an on-site or an off-site telephone.
- 3 Page through the screens until you see the Extended Call Fwd Activate Busy D/A All field.
- 4 In the Extended Call Fwd Activate Busy D/A All field, type the FAC to activate Call Forwarding from an on-site or an off-site telephone.

- 5 In the Call Fwd Activate Busy D/A All Deactivation field, type the FAC to deactivate Call Forwarding from an on-site or an off-site telephone.

- 6 Press **Enter** to save your changes.

The system displays the FACs only if the Cvg Of Calls Redirected Off-Net field on the *Optional Features* screen is set to **y**

To assign a Class of Service (COS) for extended forwarding:

- 1 Type **change cos**. Press **Enter**.

The system displays the *Class of Restriction* screen ([Figure 171, Class of Restriction screen](#), on page 645).

Figure 170: Class of Service screen

change cos	CLASS OF SERVICE															Page 1 of 1	
	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 (PSA)	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	

- 2 In the Extended Forwarding All field, type **y** in the column of each COS that allows a user to use extended Call Forwarding.
- 3 In the Extended Forwarding B/DA field, type **y** in the column of the COSs that allow a user to use extended Call Forwarding Busy/Don't Answer.
- 4 Press **Enter** to save your changes.

To assign a COR to change coverage from an on-site or an off-site telephone:

- 1 Type **change cor n**, where **n** is the number of the COR to which you want to assign a COR to change coverage from an on-site or an off-site telephone. Press **Enter**.

The system displays the *Class of Restriction* screen ([Figure 171, Class of Restriction screen](#), on page 645).

Figure 171: Class of Restriction screen

change cor 1	CLASS OF RESTRICTION	Page 1 of 4
COR Number: 1 COR Description:		
FRL: 7 Can Be Service Observed? y Can Be A Service Observer? y Partitioned Group Number: 1 Priority Queuing? n Restriction Override: none Restricted Call List? n Access to MCT? y Group II Category For MFC: 7 Send ANI for MFE? n MF ANI Prefix: Hear System Music on Hold? y APLT? y Calling Party Restriction: none Called Party Restriction: none Forced Entry of Account Codes? y Direct Agent Calling? y Facility Access Trunk Test? n Can Change Coverage? y Fully Restricted Service? n Hear VDN of Origin Annc.? y Add/Remove Agent Skills? y Automatic Charge Display? n PASTE (Display PBX Data on Phone)? y Can Be Picked Up By Directed Call Pickup? n Can Use Directed Call Pickup? n Group Controlled Restriction: inactive		

- 2 In the Can Change Coverage? field, perform one of the following actions:
 - Type **y** if you want users to change coverage from an on-site or an off-site telephone.
 - Type **n** if you do not want users to change coverage from an on-site or an off-site telephone.
- 3 Press **Enter** to save your changes.

Screens for administering Extended User Administration of Redirected Calls

Screen name	Purpose	Fields
<i>Feature Access Code (FAC)</i>	Specify the FAC for extended administration of Call Forward Busy/Don't Answer and Call Forward All Calls.	- Extended Call Fwd Activate Busy D/A All - Extended Call Fwd Activate Busy D/A All Deactivation
	Specify the FAC for extended administration of the lead-coverage path.	Change Coverage

Screen name	Purpose	Fields
<i>Class of Restriction</i>	Administer a COR that allows the use of extended administration of the lead-coverage path.	Can Change Coverage
<i>Class of Service</i>	Administer a Class of Service (COS) that allows the use of extended administration of Call Forward Busy/Don't Answer and Call Forward All Calls.	- Extended Forwarding All - Extended Forwarding B/DA
<i>Optional Features</i>	Enable Extended User Administration of Redirected Calls.	Cvg Of Calls Redirected Off-Net
<i>Station</i>	Specify a station security code (SSC) for a user.	- Security Code - Coverage Path 1 - Coverage Path 2
<i>Telecommuting Access</i>	Define a telecommuting access extension for the system.	Telecommuting Access Extension

Assigning a station security code to a user

To assign a station security code (SSC) to a user:

- 1 Type **change station *n***, where *n* is the number of the user extension to which you want to assign an SSC. Press **Enter**.

The system displays the **Station** screen, ([Figure 172, Station screen](#), on page 647).

Figure 172: Station screen

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?	
Model:		
Survivable GK Node Name:	Media Complex Ext:	
	IP Softphone? y	
	Remote Office Phone? y	

- 2 In the **Security Code** field, type the SSC.
The **Minimum Security Code Length** field on the *Feature-Related System Parameters* screen determines the length of the SSC.
The user must have a Coverage Path 1 and a Coverage Path 2. For more information, see the “Call Coverage” feature.
- 3 Press **Enter** to save your changes.

End-user procedures for Extended User Administration of Redirected Calls

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Extended User Administration of Redirected Calls:

- [Changing Call Coverage or activating Call Forwarding from an on-site telephone that is not assigned to the user](#)
- [Deactivating Call Forwarding from an on-site telephone that is not assigned to the user](#)

Changing Call Coverage or activating Call Forwarding from an on-site telephone that is not assigned to the user

To change Call Coverage or to activate Call Forwarding from an on-site telephone that is not assigned to the user:

- 1** Perform one of the following actions:
 - Enter the Change Coverage Feature Access Code (FAC) to change the Call Coverage path
 - Enter the Extended Call Fwd Activates Busy D/A All FAC to change Call Forward Busy Don't Answer

The system generates a dial tone to prompt you for the extension.

- 2** Perform one of the following actions:
 - Enter the telephone extension and then a pound sign (#), if the telephone has only bridged appearances
 - Enter the pound sign (#), to indicate that the system must use the extension of the currently active bridged appearance

- 3** Enter the station security code (SSC) for the extension, and then a pound sign (#)

The system generates a dial tone to prompt you for the forwarded-to extension

- 4** Enter the forwarded-to extension

The system generates a confirmation tone to indicate that the forwarded-to extension is valid.

- 5** Enter the digit 1 or 2 to indicate that the change request is for the first or the second coverage option, respectively

The system generates a confirmation tone to indicate that the coverage is changed.

Deactivating Call Forwarding from an on-site telephone that is not assigned to the user

To deactivate Call Forwarding from an on-site telephone:

- 1** Enter the Extended Call Fwd Activates Busy D/A All Deactivation FAC

The system generates a dial tone to prompt for the extension.

- 2** Perform one of the following actions:
 - Enter the telephone extension and then a pound sign (#), if the telephone has only bridged appearances
 - Enter the pound sign (#), to indicate that the system must use the extension of the currently active bridged appearance

If the user is at a BRI telephone with a display, the system displays all the information that the user enters, until the user completes the command sequence to change Call Coverage or to activate Call Forwarding.

If the user is at a telephone with a display, but the telephone is not a BRI telephone, the pound sign (#) is the last character the system displays, until the user completes the command sequence to change Call Coverage or to activate Call Forwarding.

- 3** Enter the SSC for the extension and then a pound sign (#).

The system generates a dial tone to prompt for the forwarded-to extension

Reports for Extended User Administration of Redirected Calls

The following reports provide information about the Extended User Administration of Redirected Calls feature:

- None

Considerations for Extended User Administration of Redirected Calls

This section provides information about how the Extended User Administration of Redirected Calls feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Extended User Administration of Redirected Calls under all conditions. The following considerations apply to Extended User Administration of Redirected Calls:

- None

Interactions for Extended User Administration of Redirected Calls

This section provides information about how the Extended User Administration of Redirected Calls feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Extended User Administration of Redirected Calls in any feature configuration.

- Bridged Appearance

When a user presses the pound sign (#) from a bridged appearance immediately after the user enters any of the Feature Access Codes (FACs) for Extended User Administration of Redirected Calls, the system allows the user to administer the extension of the currently active bridged extension. After the user enters the initial pound sign (#), the user must enter the station security code (SCC) of the currently active bridged extension to complete the command sequence.

If the station has bridged appearances only, the user must dial a station extension after the FAC to complete the command sequence.

- Call Coverage

Users can use the Extended User Administration of Redirected Calls feature to change the lead coverage path.

The system denies an attempt to activate Send All Calls, if the coverage criteria of the currently active coverage path does not allow Send All Calls.

If a user activates Send All Calls, and then changes the coverage path to a path that does not allow Send All Calls, the Send All Calls button remains lit. If the user changes the coverage path back to a path that allows Send All Calls, Send All Calls is automatically available to the user.

- 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 Calls and Call Forward Busy/Don't Answer buttons for that extension are extinguished.

The system does not support forwarding to an off-network location.

- Distributed Communications System (DCS)

You must assign a different telecommuting access extension for each server. Users can use Extended User Administration of Redirected Calls from any of the DCS nodes. The user must enter the extension of the node on which the telephone of the user is defined before the user enters the FAC.

- Security Violation Notification (SVN)

If you enable SVN, the system tracks and reports Extended User Administration of Redirected Calls security violations for SSCs.

- Tenant Partitioning

The system denies the request if the tenant number of the extension that a user dials is not accessible by the tenant number from which the user dials an FAC for Extended User Administration of Redirected Calls. If the user is at an on-site telephone when the user enters the FAC, the tenant number of the telephone from which the user enters the FAC must have access to the tenant number of the extension that the user enters. If the user dials the FAC from an off-site telephone, the tenant number of the incoming trunk must have access to the tenant number of the extension that the user dials.

Facility and Non-Facility Associated Signaling

Use Facility Associated Signaling (FAS) to allow an ISDN-PRI T1/E1 interface D-channel to carry signaling information for all the bearer (B) channels on its associated span.

Use Non-Facility Associated Signaling (NFAS) to allow one ISDN-PRI T1/E1 interface D-channel to carry signaling information for as many as 300 bearer (B) channels on its associated spans.

Detailed description of Facility and Non-Facility Associated Signaling

This section provides a detailed description of the Facility and Non-Facility Associated Signaling feature.

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 one 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 single D-channel can carry signaling information for numerous B-channels that are located on different DS1 circuit packs.

NOTE:

NFAS is valid for T1/E1 Country Protocol 1 only. Digital T1 service is also sometimes called DS1 to distinguish the service from analog T1 service.

ISDN-BRI trunks do not support NFAS.

For more information, see the “ISDN” feature.

D-channel backup with NFAS

With NFAS, you can administer a backup D-channel to improve reliability. The system switches to the backup D-channel, if a signaling link failure occurs on the primary D-channel span.

You administer one D-channel as the primary D-channel, and another D-channel as the secondary D-channel. These assignments ensure that both D-channels are in the same state at the same time, and that neither channel can be used to carry B-channel traffic at any time. The primary D-channel has precedence over the secondary D-channel.

When D-channel backup is activated, the system preserves all calls that are answered. However, some call-related information can be lost. Calls that are not answered when D-channels are switched, can also lose call-related information.

[Figure 173, ISDN-PRI configuration](#), on page 652 shows a possible configuration that involves three ISDN-PRIs between a DEFINITY Server and another DEFINITY Server or the public network.

Figure 173: ISDN-PRI configuration

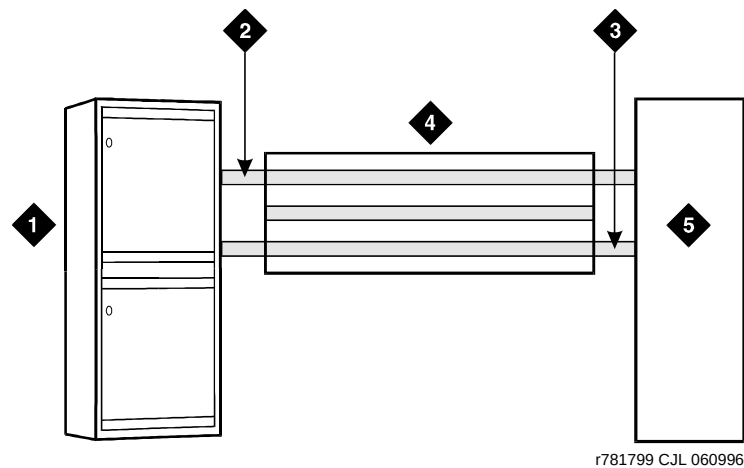


Figure notes

1	Avaya Media Server	4	ISDN-PRI controlled by D-channel
2	Secondary D-channel	5	Network far-end DEFINITY server
3	Primary D-channel		

With T1 (24-channel) interfaces, two of the ISDN-PRIs contain a D-channel and 23 B-channels. 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 carry 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 back up 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 that carries the D1 channel 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 the hardware on D1 recovers from the failure, D1 becomes the standby D-channel.

- System Technician

If a system technician sends a command to switch over a D-channel, the system tears down the signaling link on D1. Then, the system sends a message on D2 to request that D2 become the active D-channel. D2 then becomes the active D-channel, and the switchover is complete.

Hardware requirements for Facility and Non-Facility Associated Signaling

The Facility and Non-Facility Associated Signaling feature requires the following hardware:

- None

Administering Facility and Non-Facility Associated Signaling

The following steps are part of the administration process for the Facility and Non-Facility Associated Signaling feature:

- [Reviewing the guidelines for FAS and NFAS](#)
- [Implementing FAS and NFAS](#)

This section describes:

- Any prerequisites for administering the Facility and Non-Facility Associated Signaling feature
- The screens that you use to administer the Facility and Non-Facility Associated Signaling feature
- Administration procedures for the Facility and Non-Facility Associated Signaling feature

Prerequisites for administering Facility and Non-Facility Associated Signaling

You must complete the following actions before you can administer the Facility and Non-Facility Associated Signaling feature:

- None

Screens for administering Facility and Non-Facility Associated Signaling

Screen name	Purpose	Fields
<i>DS1 Circuit Pack</i>	Define the signaling mode.	Signaling Mode
<i>Interface Links</i>	Create an association between the D-channel on a DS1 circuit pack and the port on the TN765 Processor interface used for this link.	All

Screen name	Purpose	Fields
<i>Processor Channel</i>	Assign processor channels to the link that is administered on the <i>Interface Links</i> screen.	All
<i>Signaling Group</i>	Define the signaling group.	All
<i>Trunk Group</i>	Add trunk ports to the trunk group, and to the signaling group.	• Port • Sig Grp

Reviewing the guidelines for FAS and NFAS

To review the guidelines for Facility Associated Signaling (FAS) and Non-Facility Associated Signalling (NFAS):

- 1 Decide which T1/E1 facilities use FAS.
- 2 Decide which of the remaining T1/E1 facilities carry D-channel signaling information on the sixteenth (E1) or the twenty-fourth (T1) channel. For those channels that have a D-channel backup, D-channel pairs must be allocated.
- 3 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 an NFAS Signaling Group.
 - A FAS Signaling Group must contain all the ISDN B-channels on the T1/E1 interface that are associated with the D-channel of the group. An FAS signaling group 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 instead of an ISDN trunk group. These tie trunks cannot be members of a Signaling Group.
 - No restriction exists 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 the B-channels of a T1/E1 interface, (ports 1–12, for example), the group is considered to be an NFAS Signaling Group, not an FAS Signaling Group. The remaining B-channels on the T1/E1 interface are then assigned as members of another NFAS Signaling Group.
- 4 You must assign an Interface ID to each T1/E1 facility in an NFAS Signaling Group. For example, if the B-channels in a Signaling Group span three T1/E1 facilities, a unique Interface ID must be assigned to each of the three 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 by both sides of the interface, and administered before initialization.
- 5 Primary and secondary D-channel backup must be agreed by both sides of the interface, and administered before initialization. If the IDs do not match, the signaling group comes up, but calls fail.

Implementing FAS and NFAS

To implement FAS and NFAS, you must administer the following screens, in the order listed:

- 1 Administer the **DS1 Circuit Pack** screen.
- 2 Administer the **Interface Link** screen and associated screens.

You can administer the **Interface Link** screen and associated screens any time after you administer the **DS1 Circuit Pack** screens, with the following restrictions:

- You cannot assign a D-channel on a **Signaling Group** screen, unless the associated link is disabled.
- You cannot assign a trunk member until you administer the associated Signaling Group.

- 3 Administer the **ISDN-PRI Trunk Group**, **Signaling Group**, and **Trunk Group** screens.

The screens in this administration section show the DS1 interface configuration for NFAS.

Administering the DS1 Circuit Pack screen

To administer the **DS1 Circuit Pack** screen:

- 1 You must specify the **Signaling Mode** field for each DS1 circuit pack. Because the circuit pack in [Figure 174, DS1 Circuit Pack screen](#), on page 655 shows 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.

Figure 174: DS1 Circuit Pack screen

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

Administering the Interface Links and Processor Channels screens

To administer the **Interface Links** and **Processor Channels** screens:

- 1 Complete the **Interface Links** and **Processor Channels** screens for the ISDN-PRI interface on DEFINITY SI configurations, if the D-channel is switched through the TN765 Processor Interface (PI) circuit pack:
 - a Use the **Interface Links** screen to create an association between the D-channel on a DSI circuit pack, and the port on a TN765 Processor Interface circuit pack that is used for this link.
 - b Use the **Processor Channels** screen to assign processor channels to the link that you administered on the Interface Links screen.

Administering the ISDN-PRI Trunk Group, Signaling Group, and Trunk Group Members screens

To administer the ISDN-PRI Trunk Group, Signaling Group, and Trunk Group Members screens:

- 1 Note the following details shown in [Figure 175, Signaling Group screen \(Group 1\) — D-channel backup, Three DS1 Interfaces](#), on page 656, [Figure 176, Signaling Group screen \(Group 2\) — No D-channel backup, Two DS1 Interfaces](#), on page 656, and [Figure 177, Signaling Group screen \(Group 3\) — Facility Associated Signaling](#), on page 657:
 - Signaling Group 1 B-channels on DS1 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 situation. Note that the system does not display the Secondary D-channel and Trunk Board/Interface ID fields when the Associated Signaling field is set to y.

Figure 175: Signaling Group screen (Group 1) — D-channel backup, Three DS1 Interfaces

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:	—

Figure 176: Signaling Group screen (Group 2) — No D-channel backup, Two 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:	—

Figure 177: Signaling Group screen (Group 3) — Facility Associated Signaling

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: ____			

- 2 In the Sig Grp column on the [Figure 178, ISDN-PRI Trunk Group screen — Trunk Members with Required Signaling Group](#), on page 657 perform the following actions:
 - If a DS1 interface appears in one, and only one, Signaling Group, leave the Sig Grp field 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, type the Signaling Group numbers in the appropriate fields.
- 3 Press **Enter** to save your changes.

Figure 178: ISDN-PRI Trunk Group screen — Trunk Members with Required Signaling Group

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: _____					—	

Reports for Facility and Non-Facility Associated Signaling

The following reports provide information about the Facility and Non-Facility Associated Signaling feature:

- None

Considerations for Facility and Non-Facility Associated Signaling

This section provides information about how the Facility and Non-Facility Associated Signaling feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Facility and Non-Facility Associated Signaling under all conditions. The following considerations apply to Facility and Non-Facility Associated Signaling:

- None

Interactions for Facility and Non-Facility Associated Signaling

This section provides information about how the Facility and Non-Facility Associated Signaling feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Facility and Non-Facility Associated Signaling in any feature configuration.

- None

Facility Restriction Levels

Use the Facility Restriction Levels (FRL) feature to restrict some types of calls to specific users. For example, you can use FRL to allow some users to place international calls, but allow other users to place only local calls.

Facility Restriction Levels supports the following capabilities:

- **Alternate Facility Restriction Levels (AFRL)**
Use AFRL to assign a second set of FRLs within a route pattern or to lines and trunks. For example, you can use an AFRL to disable the ability to place a long distance calls when the office is closed.
- **Traveling Class Marks (TCM)**
TCM allows the system to pass the FRL of a caller from one server to another server. The server that receives the FRL uses the FRL to determine the calling privileges that are assigned to the user.

Detailed description of Facility Restriction Levels

This section provides a detailed description of the Facility Restriction Levels (FRL) feature.

The FRL controls the privileges of the call originator. The system compares the FRL of the call originator with the FRL of the call termination point. The system allows the call to continue if the FRL of the call originator is equal to or greater than the FRL of the:

- Trunk group that is the terminating point of a call that is not an automatic alternate routing (AAR) or an automatic route selection (ARS) call
- Route pattern that is assigned to the trunk that is the terminating point of an AAR or an ARS call

AAR and ARS calls

Originators of AAR and ARS calls

An originator of an AAR or an ARS call can be:

- An attendant
- A telephone user
- A remote access user
- A data terminal with a keyboard
- An incoming tie trunk group from a subtending location
- An incoming intertandem tie-trunk group, at a server or a switch
- An incoming access tie trunk group that links a remote main server or switch to a tandem server or a switch

When the system determines the FRL of the call originator, the system uses the FRL that is assigned to the COR of:

- A telephone user
- All incoming tie trunk groups
- An attendant group for attendant-extended calls
- The individual attendant, if Individual Attendant Access is assigned
- The data module that is associated with a data terminal
- The barrier code that a user dials for a remote access call

If the remote access call does not require a barrier code, no FRL exists.

Call termination points for AAR and ARS calls

A termination point for an AAR or an ARS call can be:

- A tie trunk
 - A tie trunk termination point for an AAR and ARS call can include a common-control switching arrangement (CCSA) access trunk and an enhanced private switched communications services (EPSCS) access trunk.
 - A tie trunk termination point for an AAR and ARS call does not include a release-line trunk (RLT).
- A Wide Area Telecommunications Services (WATS) trunk
- A central office (CO) trunk
- A foreign exchange (FX) trunk
- An integrated services digital network-primary rate interface (ISDN-PRI) trunk

Each of these outgoing trunk groups has a COR that contains an FRL. However, for AAR and ARS calls, the system uses the FRL that you assigned to the route pattern of the trunk group.

AFRL

Alternate Facility Restriction Levels (AFRL) allows you to define a second set of facility restriction levels within a route pattern, or for lines or trunks. Attendants and system administrators can activate the AFRLs and change user access to lines and trunks. For example, you can use AFRL to disable the ability to place a long distance call when the office is closed.

AFRL alters the route patterns for originating telephones, originating trunks, and dialed authorization codes. If AFRL is active:

- TCM is set to a new FRL value
- The TCM information that the system records in the Call Detail Recording (CDR) records is the value of the AFRL, not the original TCM.



CAUTION:

AFRL has an impact on both AAR and ARS call routing because AFRL can change routing preferences. The use of AFRL on tandem and tie-trunk applications can affect entire networks. The system can block calls that are part of a cross-country private network that need to be routed further.

Alt-frl feature button

You can assign an **alt-frl** button to any attendant console and to any user telephone. The attendant or the user presses the **alt-frl** button to activate and deactivate the AFRL. The use of the **alt-frl** button can affect the status of other buttons.

When AFRL is active, the user might notice a change in calling privileges. Consider notifying users of the changes in calling privileges, and prepare your telecommunications department to respond to user inquiries.

Authorization codes

Authorization codes prevent unauthorized access to some system facilities. When a user dials an authorization code, the system checks the code. If the code is invalid, the system generates the intercept tone. If the code is valid, the system uses the COR and the FRL that is associated with the authorization code for further call processing. However, if AFRL is activated, the system uses the AFRL for further call processing.

If the system uses an intertandem tie trunk group for a call, the system outpulses a TCM as the last digit of the number. If the FRL of the intertandem tie-trunk is equal to or greater than the terminating FRL, the system proceeds with call processing. If the FRL of the originator is less than the FRL of the termination point, the system compares the TCM with the FRL of the tie-trunk. If the TCM is greater than or equal to the FRL of the tie trunk, the system proceeds with call processing.

Hardware requirements for Facility Restriction Levels

The Facility Restriction Levels (FRL) feature requires the following hardware:

- None

Administering Facility Restriction Levels

This section describes the screens that you use for administering the Facility Restriction Levels (FRL) feature.

Screens for administering Facility Restriction Levels

Screen name	Purpose	Fields
AAR Digit Analysis Table	Associate a dialed string with a route pattern, and thus to an FRL.	Route Pattern
ARS Digit Analysis Table	Associate a dialed string with a route pattern, and thus to an FRL.	Route Pattern
Attendant Console	Assign an alt-frl button for the attendant so that the attendant can activate the AFRL capability.	alt-frl
	Assign a COR for the attendant to associate an FRL with the attendant.	COR
Class of Restriction	Assign an FRL to the Class of Restriction (COR).	FRL
Console Parameters	Assign a COR for the attendant group to associate an FRL with the attendant group.	COR
Data Module	Assign a COR for the data module to associate an FRL with the data module.	COR
Remote Access	Assign a COR to the barrier code to associate an FRL with the barrier code.	COR
Route Pattern	Assign an FRL to the trunk group.	FRL
Station	Assign an alt-frl button for a user so that the user can activate the Alternate Facility Restriction Levels (AFRL) capability.	alt-frl
	Assign a COR for the user to associate an FRL with the user.	COR
Trunk Group	Require that a user enter an authorization code, if the user wants to tandem a call through an Automatic Alternate Routing (AAR) or an Automatic Route Selection (ARS) route pattern.	Auth Code
	Assign a COR for the trunk group to associate an FRL with the trunk group.	COR

End-user procedures for Facility Restriction Levels

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Facility Restriction Levels (FRL):

- Using the Alternate Facility Restriction Levels (AFRL) capability

To use the AFRL capability:

- To activate AFRL, press the alt-frl button.
- To deactivate AFRL, press the alt-frl button.

Reports for Facility Restriction Levels

The following reports provide information about the Facility Restriction Levels feature:

- None

Considerations for Facility Restriction Levels

This section provides information about how the Facility Restriction Levels (FRL) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Facility Restriction Levels under all conditions. The following considerations apply to Facility Restriction Levels:

- Trunk groups
 - Use the **Route Pattern** screen to assign the FRL to a trunk group.
 - You can use the same trunk group in more than one route pattern.
 - 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.
- General access

Be consistent in FRL assignments. Always use FRL 0 or 1 for a trunk group that everyone can access.
- Route patterns

If you use a range of 0 through 5 in one route 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 the users to local calls. Use any other number for the FRL on your first-choice route pattern.
- Remote access barrier codes

You assign FRLs for remote access users through the remote-access barrier codes. The simplest way to assign these FRLs is to duplicate the on-premises FRLs, and then relate the appropriate barrier code to users who use remote access. For more information, see the “Remote Access” feature.

Interactions for Facility Restriction Levels

This section provides information about how the Facility Restriction Levels feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Facility Restriction Levels in any feature configuration.

- **Call Detail Recording (CDR)**

If your system uses a 15-digit CDR account, the system overwrites the FRL field in the CDR record with the account code.

Feature Access Codes

Use Feature Access Codes (FAC) to provide users with quick access to certain features of the telephone system. When you assign a FAC to a feature, users do not have to program a button on the telephone to use this feature. Instead, users just dial the FAC.

Detailed description of Feature Access Codes

This section provides a detailed description of Feature Access Codes.

Feature Access Codes can contain from one to four characters. These characters can be all digits, or a combination of digits and either an asterisk (*) or a pound sign (#). If you use an asterisk (*) or a pound sign (#), these characters must appear in the first position of the FAC. Before you define a FAC that includes an asterisk or a pound sign, consider that users with analog rotary dial telephones cannot dial these codes.

The asterisk and pound sign are often used in pairs. Use one character plus digits to activate a feature, and the other character plus the same digits to deactivate that feature. For example, say that you use the asterisk and then the digits 2 and 9 to activate the Posted Messages feature. You might then use the pound sign and the same digits 2 and 9 to deactivate the Posted Messages feature.

FACs are divided into four types:

- Access FACs
An access FAC gives a user access to a feature.
- Activate or Deactivate FACs
An activate and a deactivate pair of FACs allows a user to activate or deactivate a specific feature.
- Send or Cancel FACs
A send and a cancel pair of FACs allows a user to send or cancel a message.
- Lock or Unlock FACs
A lock and an unlock pair of FACs allows a user to lock or unlock the message retrieval capability of a telephone.

Many features already have default FACs assigned. You can use these default FACs, or you can change the default FACs to codes that make more sense to you.

Each FAC must conform to your dial plan, and each FAC must be unique. If you try to define a FAC that is already assigned to another feature, the system warns you of the duplicate FAC. You cannot continue until you change one of the FACs.

Hardware requirements for Feature Access Codes

The Feature Access Codes feature requires the following hardware:

- None

Administering Feature Access Codes

The following steps are part of the administration process for the Feature Access Codes feature:

- [Assigning Feature Access Codes](#)
- [Changing or deleting Feature Access Codes](#)

This section describes:

- Any prerequisites for administering Feature Access Codes
- The screens that you use to administer Feature Access Codes
- Complete administration procedures for Feature Access Codes

Prerequisites for administering Feature Access Codes

You must complete the following actions before you can administer Feature Access Codes:

- Ensure that Feature Access Codes are set up in your dial plan. You must have a **FAC** or **DAC** entry on the dial plan screen for the digit range that you intend to use for you Feature Access Codes. For a description of how to set up your dial plan, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Feature Access Codes

Screen name	Purpose	Fields
<i>Feature Access Code (FAC)</i>	Assign FACs to specified telephone features.	All

Assigning Feature Access Codes

To assign a Feature Access Code for a specific feature:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the *Feature Access Code (FAC)* screen ([Figure 179, Feature Access Code \(FAC\) screen](#), on page 667).

Figure 179: Feature Access Code (FAC) screen

change feature-access-codes Page 1 of 8

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: _____

Attendant 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: _____ 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: _____

Contact Closure Open Code: _____ Close Code: _____

Contact Closure Pulse Code: _____

- In the field next to the specific feature to which you want to assign the FAC, type a FAC that conforms to your dial plan. You might have to scroll through several pages of the **Feature Access Code (FAC)** screen to find the telephone feature that you want.

Some features require more than one FAC. Type a FAC in each required field. For example, type a separate FAC in the Call Forwarding Activation Busy/DA field, the All field, and the Deactivation field.
- Press **Enter** to save your changes.
- Ensure that you notify all users of the assigned FACs.

Changing or deleting Feature Access Codes

To change or delete a Feature Access Code for a specific feature:

- 1 Type **change feature-access-codes**. Press **Enter**.
The system displays the **Feature Access Code (FAC)** screen ([Figure 180, Feature Access Code \(FAC\) screen](#), on page 668).

Figure 180: Feature Access Code (FAC) screen

change feature-access-codes	Page 1 of 8
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: ____	
Attendant 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: ____ 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: ____	
Contact Closure Open Code: ____	Close Code: ____
Contact Closure Pulse Code: ____	

- 2 In the field next to the feature that you want to change, type a new FAC that conforms to your dial plan over the existing FAC. You might have to scroll through several pages of the **Feature Access Code (FAC)** screen to find the telephone feature that you want.

Some features require more than one FAC. Type a FAC in each required field. For example, type a separate FAC in the Call Forwarding Activation Busy/DA field, the All field, and the Deactivation field.
- 3 To remove any FAC, delete the existing FAC and leave the field blank.
- 4 Press **Enter** to save your changes.
- 5 Ensure that you notify all users of the changed FACs.

Reports for Feature Access Codes

The following reports provide information about Feature Access Codes:

- None

Considerations for Feature Access Codes

This section provides information about how Feature Access Codes behave in certain circumstances. Use this information to ensure that you receive the maximum benefits of Feature Access Codes under all conditions. The following considerations apply to Feature Access Codes:

- None

Interactions for Feature Access Codes

This section provides information about how the Feature Access Codes feature interacts with other features in your system. Use this information to ensure that you receive the maximum benefits of Feature Access Codes in any feature configuration. The following interactions apply to Feature Access Codes:

- None

Troubleshooting Feature Access Codes

This section lists the known or common problems that users might experience with Feature Access Codes:

Problem	Possible cause	Action
The user experiences delays when attempting to use a FAC.	You might have a FAC and an extension with the same digits on your dial plan.	Check your dial plan to see if you have a FAC and an extension with the same digits. For more information, click here , or see the <i>Administrator's Guide for Avaya Communication Manager</i> .

Feature Access Codes

Troubleshooting Feature Access Codes

Group Paging

Use the Group Paging feature to make an announcement over a group of digital speakerphones.

- You can create up to 32 paging groups on one media server.
- Each group can consist of up to 32 extensions.
- You can assign the same extension to different groups.

Detailed Description of Group Paging

With the Group Paging feature, you can create a page group, and assign extensions as members of the group. You assign an identifying extension to each page group, which users dial to page the group. When a user dials the extension of the paging group, Communication Manager activates the speakers on all the telephones in the group. Speakerphone paging is one-way communication: group members hear the person place the page, but cannot respond directly.

Restrictions

Pages are not always heard on every telephone 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. Pages cannot be heard when the Send All Calls or Do Not Disturb features are activated. See [Interactions for Group Paging](#) on page 674 for features that block group pages.

When a group member does not hear the announcement for any of these reasons, the caller is not notified. Therefore, the originator of an important page might want to check with the group members to ensure that all members heard the page.

Controlling access to paging groups

Each paging group is assigned a class of restriction (COR), so that you can provide or deny access to different classes of users by setting calling permissions appropriately. Note that you can administer CORs so that remote callers can make speakerphone pages. If you do not want to allow remote users to page, you can use the **Class of Restriction** screen to set calling permissions for vector directory numbers (VDNs) and trunk groups so that neither can initiate pages. For more information on CORs, see the “Class of Restriction” feature.

Hardware requirements for Group Paging

The Group Paging feature requires the following hardware:

- Paging group members must have Digital Communications Protocol (DCP) set speakerphones or Internet Protocol (IP) set speakerphones to use the Group Paging feature.

Administering Group Paging

The following steps are part of the administration process for the Group Paging feature:

- [Creating a Paging Group](#)
- [Changing a Paging Group](#)
- [Viewing All Paging Groups](#)

This section describes:

- Any prerequisites for administering the Group Paging feature
- The screens that are required to administer the Group Paging feature
- Complete administration procedures for the Group Paging feature

Prerequisites for administering Group Paging

You must complete the following actions before you can administer the Group Paging feature:

- None

Screens for administering Group Paging

Screen Name	Purpose	Fields
<i>Group Paging Using Speakerphones</i>	Create or change a paging group, and add or delete group members.	All
<i>Speakerphone Page Groups</i>	View a list of all existing paging groups.	All

Creating a Paging Group

To create a new paging group:

- 1 Type **add group-page *n***, where *n* is a number between **1** and **32**, or **next** for the next available group number. Press **Enter**.

The system displays the *Group Paging Using Speakerphone* screen ([Figure 181, Group Paging Using Speakerphone screen](#), on page 673).

Figure 181: Group Paging Using Speakerphone screen

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 the extension that users dial to page the members of this group. In this example, the Group Extension is 3210.
- 3 In the **Group Name** field, type the name that you want to assign to this paging group. This name appears on a the telephone display of a caller when paging the group. In this example, the Group name is **Sales staff**.
- 4 In the **COR** field, type the Class of Restriction (COR) that you want to assign to this group. Any user who wants to page this group must have permission to call COR 5.
- 5 In the **Ext** field in row 1, type the extension of the first member of the paging group. **2009**.
- 6 Type the remaining extensions that are members of this group. When you save your changes, Communication Manager automatically completes the **Name** fields with the names that are associated with the extensions on the **Station** screen.
- 7 Press **Enter** to save your changes. Paging can now be used.

Changing a Paging Group

You can add or delete members of a paging group, or modify the other attributes of the group, such as the Group Name, Group Extension, or COR.

To change a paging group:

- 1 Type **change group-page n**, where **n** is the number of the paging group that you want to change. Press **Enter**.
The system displays the **Group Paging Using Speakerphone** screen ([Figure 181, Group Paging Using Speakerphone screen](#), on page 673).
- 2 In the **Ext** field, type the extension of a member that you want to add, or delete the extension of a member that you want to remove from the group.
- 3 Make any changes to the **Group Name**, **Group Extension**, or **COR** fields.
- 4 Press **Enter** to save your changes.

Viewing All Paging Groups

To view a list of existing paging groups:

- 1 Type **list group-page**.

The system displays the *Speakerphone Page Groups* screen ([Figure 182, Speakerphone Page Groups screen](#), on page 674).

Figure 182: Speakerphone Page Groups screen

list group-page				
SPEAKERPHONE PAGE GROUPS				
Group Number	Group Extension	Group Name	COR	No. of MEMBERS
11:	696147	AUTO H-grp-page1	15	6
12:	696148	AUTO H-grp-page1	15	2
13:	696149	Test Group 1	1	4
14:	694056	Test Group 2	1	2

Reports for Group Paging

- None

Considerations for Group Paging

This section provides information about how the Group Paging feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Group Paging feature under all conditions.

- None

Interactions for Group Paging

This section provides information about how the Group Paging feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of the Group Paging feature in any feature configuration.

- Attendant Intrusion

Attendants cannot intrude on group pages. If the attendant tries to intrude on the originator of the page, 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 the coverage paths of the group members. A page group cannot be a coverage point.
- **Call Park**
Group members who receive a page cannot park the call.
- **Call Pickup and 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 servers/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.

Troubleshooting Group Paging

This section lists the known or common problems that users might experience with the Group Paging feature.

Problem	Possible cause	Action
A user gets a busy signal when the user tries to page.	All telephones in the group are busy or off-hook.	Wait a few minutes and try again.
	Send All Calls or Do Not Disturb is activated for all telephones in the group.	Group members must deactivate these features in order to hear a page.
Some group members do not hear a page.	Send All Calls or Do Not Disturb is activated for the telephones of these group members.	Group members must deactivate these features in order to hear a page.

Hold

Use the Hold feature to temporarily disconnect from a call, use the telephone for another call, and then return to the original call.

Hold supports the following capabilities:

- Automatic Hold

When an attendant or a multifunction telephone user selects a second call, the system automatically puts the active call on hold, and makes the second call active.

Detailed description of Hold

This section provides a detailed description of the Hold feature.

Multiappearance telephone users can use a Hold button to activate Hold. With Automatic Hold, a user can also press a second call appearance to put an active call on hold. The system holds the call at the call appearance that is used for the call. Multiappearance telephone users can hold a call on each call appearance.

Single-line telephone users can use two types of Hold, Soft Hold and Hard Hold.

Soft Hold

Use Soft Hold to conference or transfer a call that includes the held call. With Soft Hold, the user can put a call on hold, consult with another party, activate or deactivate a feature, and then return to the soft-held call.

Hard Hold

Use Hard Hold to perform operations that do not include the held call. The user can put a call on hold and call another party. The user can then answer a waiting call, transfer or conference the waiting call, or activate or deactivate features.

Automatic Hold

With Automatic Hold, attendants and multifunction telephone can alternate easily between two or more calls. For example, when an attendant or multifunction telephone user selects a second call, the system automatically puts the active call on hold, and makes the second call active. Automatic Hold is a system-wide capability. If you do not enable Automatic Hold for your system, the system drops the current active call when an attendant or user selects a second call. A call that is placed in Automatic Hold is in Hard Hold.

To put an active call on hold, without pressing the Hold button, the user presses a second call-appearance button. The second call appearance becomes active. A user can place more than one call on hold. However, the user must keep one call appearance available for other calls.

The controlling telephone can have only one auto-held call on soft hold at a time. A soft hold is the state of a line after the user presses a conference button or a transfer button, but before either process is complete. The controlling telephone is guaranteed the ability to reenter any auto-held call later, unless the auto-held parties disconnect, or the auto-held tone exceeds the auto-held tone time limit.

Hardware requirements for Hold

The Hold feature requires the following hardware:

- None

Administering Hold

The following steps are part of the administration process for the Hold feature:

- [Enabling Automatic Hold](#)
- [Assigning an FAC for CAS remote hold and answer](#)

This section describes:

- Any prerequisites for administering the Hold feature
- The screens that you use to administer the Hold feature
- Complete administration procedures for the Hold feature

Prerequisites for administering Hold

You must complete the following actions before you can administer the Hold feature:

- None

Screens for administering Hold

Screen name	Purpose	Fields
<i>Feature Access Code (FAC)</i>	Define the FAC for centralized attendant services (CAS) attendant remote hold and answer.	CAS Remote Hold/Answer Hold-Unhold
<i>Feature-Related System Parameters</i>	Enable the Automatic Hold capability for your system.	Auto Hold

Enabling Automatic Hold

To enable Automatic Hold for your system:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the *Feature-Related System Parameters* screen, ([Figure 183, Feature-Related System Parameters screen](#), on page 679).

Figure 183: Feature-Related System Parameters screen

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? _	
Use Trunk COR for Outgoing Trunk Disconnect? _	
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: _

- 2 Page through the screens until you see the Auto Hold field.
- 3 In the Auto Hold field, perform one of the following actions:
 - Type **y** if you want the Automatic Hold capability available to all users on your system.
 - Type **n** if you do not want the Automatic Hold capability available to any users on your system.
- 4 Press **Enter** to save your changes.

Assigning an FAC for CAS remote hold and answer

To assign a Feature Access Code (FAC) Centralized Attendant Service (CAS) remote hold and answer for your system:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the *Feature Access Code (FAC)* screen, ([Figure 184, Feature Access Code \(FAC\) screen](#), on page 680).

Figure 184: Feature Access Code (FAC) screen

change feature-access-codes	FEATURE ACCESS CODE (FAC)	Page 1 of 7
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:	_____	
Attendant 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:	_____	

- 2 In the CAS Remote Hold/Answer Hold-Unhold Access Code field, type the FAC that a Centralized Attendant Service (CAS) attendant can use to:
 - Place calls on hold
 - Answer calls that are held at a remote server that is running Avaya Communication Manager.
- 3 Press **Enter** to save your changes.

Reports for Hold

The following reports provide information about the Hold feature:

- None

Considerations for Hold

This section provides information about how the Hold feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Hold under all conditions. The following considerations apply to Hold:

- To drop a call dialed from a single-line telephone within the first 10 seconds after you complete dialing the call, flash the switch hook.
- A single-line telephone user cannot hold a call that involves an attendant. A multiappearance telephone user can hold a call that involves an attendant, unless the user attempts to conference or transfer the call.
- When only Automatic Hold is involved, and the attendant on an active loop presses a second loop, the system places the active call on Hard Hold.
- The Held Call Timed Reminder does not apply to conference calls, and is not started when a conference is placed on hold.
- Automatic Hold operates in conjunction with the START key or the Automatic Start feature of an attendant console. The START key and the Automatic Start operation have precedence over Automatic Hold, and place an active loop call on Soft Hold.

Interactions for Hold

This section provides information about how the Hold feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Hold in any feature configuration.

- **Automatic Callback**
A single-line telephone user cannot receive an Automatic Callback call while a call is on hold.
- **Bridged Call Appearance**
Any user, who is active on a bridged call, can place the call on hold. If no other users with a bridged call appearance of the same extension are connected to the call, the status lamp at the Bridged Appearance button indicates that the call is on hold. If the primary extension or another bridged appearance is connected to the call, the status lamp at all bridged appearances indicates an active status for the call.
- **Centralized Attendant Service (CAS)**
Automatic Hold does not affect the operation of CAS.
- **Distributed Communications System (DCS)**
Automatic Hold does not affect the operation of DCS, and is administered separately for each node in a DCS network.
- **Leave Word Calling (LWC)**
A multiappearance telephone user who is on hold can activate LWC toward the holding user.
A single-line telephone user cannot activate LWC toward another user while a call is on Soft Hold.

- Music-on-Hold
Only one party on hold can hear music.
- Personal Central Office Line (PCOL)
When a user, who is active on a PCOL call, puts the call on Hold, the lamp flutters or winks. The status lamp that is associated with the PCOL button lamp does not track the busy or idle status of the PCOL.
- Priority Calling
Users can receive priority ringing and have a call on soft hold.

Hot Line Service

Use the Hot Line Service feature to assign a specific destination to which the user of a single-line telephone automatically connects to a destination when the user goes off-hook.

Detailed description of Hot Line Service

This section provides a detailed description of the Hot Line Service feature.

With the Hot Line Service, the user of a single-line telephone can automatically connect to a preassigned destination when the user goes off-hook. You can assign the following numbers as a Hot Line Service destination:

- An attendant
- An extension
- A feature access code (FAC)
- A public telephone number
- A private telephone number

The Hot Line Service destination number must be stored in an Abbreviated Dialing list. When the user goes off-hook, the system automatically routes the call to the stored number, and completes the call as if the user dialed the call.

If an attendant number is the Hot Line Service destination number, the system automatically routes the telephone user to the attendant.

If the appropriate feature access code is stored with the number in the Abbreviated Dialing list, the system uses automatic alternate routing (AAR), automatic route selection (ARS), Data Privacy, or Priority Calling.

If the Public Network Access code or the Private Network Access code is the number stored in the Abbreviated Dialing list, the system connects the user to the outside number.

A Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension, or any individual extension within such a group, can be a Hot Line Service destination.

Hot Line Service does not change the way that a user receives calls. Calls to a user with Hot Line Service are controlled by the Class of Restriction (COR) that you assign to the user extension. Hot Line Service does not affect the recipient of the call.

Hardware requirements for Hot Line Service

The Hot Line Service feature requires the following hardware:

- None

Administering Hot Line Service

This section describes the screens that you use to administer the Hot Line Service feature

Screens for administering Hot Line Service

Screen name	Purpose	Fields
<i>Abbreviated Dialing List</i> • <i>System List</i> • <i>Group List</i> • <i>Personal List</i>	Store the number that you want the system to use when the user enters the dial code.	Dial Code
<i>Data Modules</i> • <i>Data Line Module</i> • <i>MPD/MTDM</i> • <i>Netcon Data Module</i> • <i>Processor Interface Data Module</i> • <i>System Port Data Module</i> • <i>Packet Gateway (PGATE)</i> • <i>Netcon Data Module</i>	Specify Hot Line Service information for the data module.	• Special Dialing Option • Hot Line Destination
<i>Station - single-line</i>	Specify Hot Line Service information for the user.	• Hot Line Destination - Abbreviated Dialing List Number • Hot Line Destination - Dial Code • Special Dialing Code

Reports for Hot Line Service

The following reports provide information about the Hot Line Service feature:

- None

Considerations for Hot Line Service

This section provides information about how the Hot Line Service feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Hot Line Service under all conditions. The following considerations apply to Hot Line Service:

- Specify the attendant as the Hot Line Service destination when you want the attendant to screen calls originations.
- You can assign Hot Line Service to any number of telephones, with the same or different destinations. The number of users who can use Hot Line Service is limited by the number of entries that you can store in the Abbreviated Dialing lists.
- A Hot Line Service user can activate any feature, but only if the access code for the feature is stored in the Abbreviated Dialing List.

Interactions for Hot Line Service

This section provides information about how the Hot Line Service feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Hot Line Service in any feature configuration.

- Night Service
When Night Service is active, the system redirects the Hot Line Service call.
- Bridged Call Appearance - Single-Line Voice Terminal
A bridged call appearance of a telephone that is administered for Hot Line Service, also places a hot line call automatically when a user goes off-hook on that bridged appearance.
- Loudspeaker Paging Access
You can use Loudspeaker Paging Access with Hot Line Service to provide automatic access to paging equipment.
- Ringback Queuing
If a Hot Line Service call accesses a trunk group with Ringback Queuing, the call can queue, unless the telephone is Termination Restricted by the Class of Service (COR). Queuing, when applicable, is automatic on single-line telephones.

Hunt Groups

Use the Hunt Groups feature to set up a group of extensions that can handle multiple calls to a single telephone number. You can choose the call distribution method to route calls. For each call to the number, the system hunts for an available extension in the hunt 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 telephone number. A hunt group might consist of people who are trained to handle calls on specific topics. For example, the group might be a:

- Benefits department within your company
- Service department for products you sell
- Travel reservations service
- Pool of attendants

A hunt group might also consist of a group of shared telecommunications facilities. For example, the group might be a:

- Modem pool
- Group of data-line circuit ports
- Group of data modules

The Hunt Groups feature supports the following capabilities:

- Announcements for hunt groups
When an extension is not immediately available, the system plays a recorded announcement to the caller.
- Call Coverage for hunt groups
You can set up a call coverage plan so that when no hunt group extension is available, calls go to a coverage point.
- Call Distribution methods for hunt group types
You can specify one of the following methods to determine how the system distributes calls members of the hunt group:
 - Direct Department Dialing (DDD)
 - Uniform Call Distribution (UCD)
 - Circular
 - Automatic Call Distribution (ACD)
 - Expert Agent Selection (EAS)
- Hunt group extension unavailability
A hunt group extension that is occupied on another call is unavailable to take a new call to the hunt group number. Use of Hunt Group Busy, Send All Calls, or Call Forwarding All Calls buttons also makes a hunt group member unavailable for calls.
- Queues for hunt groups
You can set up a queue for hunt group calls that cannot be answered immediately.
- Teletypewriter (TTY) hunt groups
You can create a hunt group that includes TTY-equipped agents to accommodate hearing-impaired callers.

Detailed description of Hunt Groups

This section provides a detailed description of the Hunt Groups feature.

Announcements for hunt groups

You can record and assign one delay announcement to each hunt group queue. An announcement can be shared among hunt groups. Usually, a hunt group announcement asks the caller to wait, and says that the call will be answered in the order in which the call was received.

A call that connects to a delay announcement remains in a queue while the system plays the announcement. If the call is not answered by the time that the announcement completes, the caller hears music, if music is administered. If music is not administered, the caller hears silence. When the call starts ringing at the telephone of a hunt group member, the caller hears ringing.

Delay announcement intervals for hunt groups

You can define a delay announcement interval for each hunt group. This interval of from 0 to 99 seconds specifies how long a call remains in the queue before the system connects the call to a recorded announcement. When a call enters the queue, the interval starts. If Call Coverage is administered, the Don't Answer interval of from one to 99 ringing cycles also starts when the call enters the queue. After these intervals start, one of the following processes also starts:

- If the Don't Answer interval expires before the delay announcement interval expires, the system redirects the call to coverage.
- If no coverage point is available, the call remains in the queue and the system connects the call to the delay announcement when the delay announcement interval expires.
- If the delay announcement interval expires before the Don't Answer interval, the system connects the call to a delay announcement. If the announcement is already in use, the system resets the delay announcement interval.

This process continues as above 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, the system automatically connects a call to the announcement. This announcement is called a *forced first announcement*. In this case, the system does not connect the call to a hunt group member until after the announcement. The caller does not hear music.

If the system redirects a call to another hunt group through Call Coverage, the caller does not hear the forced first announcement of either hunt group. However, the caller might hear the first or the second announcement of the covering hunt group.

Analog, aux-trunk, or integrated delay announcements for hunt groups

Delay announcements can be analog, aux-trunk, or digitally integrated. For an analog or aux-trunk announcement, a caller who enters the queue hears the associated announcement the next time that the system plays the announcement. A caller who enters the queue after the announcement starts does not hear the announcement until the announcement starts again. For an integrated announcement, multiple callers can be connected to the same announcement at different times, depending on the available ports.

Example

Assume that a hunt group has the following parameters:

- The queue length is 10 calls.
- The queue warning level is 5 calls.
- The 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 current calls. Meanwhile, the call waits in the queue for 20 seconds, and the system plays the recorded announcement. When a hunt group member becomes available, the first call in the queue connects to that group member. The queue warning-level lamp turns off when the number of calls in the queue falls to four.

Call Coverage for hunt groups

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, the call stays in the queue for the Don't Answer interval. The system then redirects the call to the coverage point. A call coverage point can be another hunt group.

For more information on setting up call coverage, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Call Distribution methods for hunt group types

The system can use different types of station hunting methods to distribute calls to hunt groups. You specify the call distribution method in the **Group Type** field on the **Hunt Group** screen. The available values for the **Group Type** field depend on how your system is configured.

You have more call distribution options if your company acquires Automatic Call Distribution (ACD) or Expert Agent Selection (EAS). These options are available if the **ACD** and **Expert Agent Selection (EAS)** fields on the **Optional Features** screen are enabled. ACD and EAS distribute calls according to the workloads and the skill levels of the agents in each hunt group. You can use this type of call distribution to track call handling and monitor the efficiency of agents. When you assign ACD to a hunt group, the group is called a *split*. When you assign EAS to a hunt group, the group is called a *skill*. For more information on Expert Agent Selection, see the *Avaya Communication Manager Call Vectoring and Expert Agent Selection Guide*. For more information on ACD and multiple call handling, see the *Avaya Communication Manager Contact Center Guide to ACD Contact Centers*.

The following table lists the different hunt group types and how each type handles incoming calls.

Group type	Hunting method	Extra software needed
circ (Circular)	The system routes calls in a “round-robin” order. The order in which participating extensions are administered is the order in which calls are directed. The system tracks the last extension in the hunt group to which a call was connected. The next call to the hunt group is directed to the next extension on the list, regardless of idle time.	None
ddc (Direct Department Calling)	Direct Department Calling is also known as “hot seat” distribution. The system starts with the first extension in the hunt group, and hunts for an available extension. If the first extension is busy, the system checks the second extension. If the second extension is busy, the system checks the third extension, and so on. When an extension is available, the system 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. When the group is administered as a skill, ddc is unavailable.	None
ead-loa (Expert Agent Distribution - Least Occupied Agent)	The system hunts for the available agent who has the highest skill level, and the lowest percentage of work time since the agent logged in.	EAS and CentreVu Advocate
ead-mia (Expert Agent Distribution - Most Idle Agent)	The system hunts for the available agent who has the highest skill level, and the longest idle time since the last call.	EAS
pad (Percent Allocation Distribution)	The system selects from a group of available agents based on a comparison of work time in the skill, and the target allocation for the skill.	
slm (Service Level Maximizer)	The system compares the current skill level for each administered skill to a user-defined call service level target. The system selects only those agents whose other skills have the least need for their service at the current time.	

Group type	Hunting method	Extra software needed
ucd-loa (Uniform Call Distribution - Least Occupied Agent)	The system hunts for the agent with the lowest percentage of work time since the agent logged in.	ACD, EAS, and Avaya Business Advocate
ucd-mia (Uniform Call Distribution - Most Idle Agent)	The system hunts for the agent who has been idle the longest since their last call.	None

Hunt group extension unavailability

An extension in a hunt group is unavailable to receive calls if the hunt group member is already handling another call. This rule is true even if the call is not a hunt group call, and even if the telephone is a multi-appearance telephone.

An extension also becomes unavailable if the member presses one of the following buttons:

- Hunt Group Busy, or if the member enters the Hunt Group Busy Activate feature access code (FAC)
- Send All Calls
- Call Forwarding All Calls

Hunt Group Busy option

A hunt group member can dial the Hunt Group Busy code and then the hunt group number to make the extension of the member unavailable for calls. The extension remains unavailable until the group member dials the Hunt Group Busy deactivation code.

If the last available member of a hunt group tries to activate the Hunt Group Busy option, the following events occur:

- New calls to the hunt group receive a busy tone or go to coverage.
- The system continues to route calls that are already in the queue to the last available extension.
- When the queue is empty, the system activates Hunt Group Busy. At the last available extension, if a status lamp is associated with the **Aux Work** button, the lamp flashes until the queue is empty. When no more calls remain in the queue, Hunt Group Busy activates and if a status lamp is provided, the lamp lights steadily, but does not flash.

If an agent is an ACD split and a hunt group member, the agent in the split usually has an **AUX-work** button that also activates/deactivates Hunt Group Busy. If an agent is the last available member and pushes **AUX-work**, the lamp on the button flashes until the queue is empty. The flashing light means that the agent is still available. When the queue is empty, the lamp lights but does not flash, and Hunt Group Busy takes effect.

Using Send All Calls with hunt groups

If a station activates Send All Calls with the **Send All Calls** button, hunt group calls go in the queue, if a queue is administered. If a queue is not administered, callers get a busy treatment if Send All Calls is activated for all the agents.

If an extension is an agent in an ACD split and a hunt group member, the split agent usually has an **AUX-work** button that also activates and deactivates Hunt Group Busy. If an agent presses the **Send All Calls** button, the agent becomes unavailable for hunt group calls. The agent becomes available for hunt group calls again when the agent presses the **Send All Calls** button again.

Using Call Forwarding All Calls with hunt groups

With Call Forwarding All Calls active, an extension within a hunt group is unavailable for hunt group calls. If a forced first announcement is administered for the hunt group, callers hear the forced first announcement before the system forwards the call.

Queues for hunt groups

You can set up a queue for a hunt group. When all extensions in the group are busy, calls wait in the queue for the next available extension. You set the length of the queue to determine how many calls can wait in queue.

If all hunt group members are unavailable or the queue is full, the system treats the call in one of the following ways:

- If the call is internal or is carried on a DID (Direct Inward Dialing), DS1 (Digital Signal Level 1), or tie trunk, the caller hears a busy tone.
- If the call is on a central office (CO) 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.

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 all members of the hunt group can see the lamp.

TTY hunt groups

Several laws require that “reasonable accommodation” be provided for people with disabilities. For this reason, your company might offer support for callers who use teletypewriters (TTYs).

To accommodate TTY callers, you can create a hunt group that includes agents who are equipped with a TTY. Many TTYs can connect directly to the telephone network by means of analog RJ-11 jacks. However, Avaya recommends that agents be equipped with TTYs that include an acoustic coupler that can accommodate a standard telephone handset. One reason for this recommendation is that a large proportion of TTY users are hearing impaired, but speak clearly. These people often prefer to receive calls on the TTY, and then speak in response. The call center agent must alternate between listening on the telephone and typing on the TTY. This process is easier with an acoustically coupled configuration.

Although TTY-emulation software packages are available for personal computers, few of these can intermix voice and TTY on the same call.

For a TTY hunt group, you can record TTY announcements and use announcements for the hunt group queue. To record announcements for TTY, follow the same procedure that you use for voice recordings from your telephone. However, instead of speaking into your telephone to record, type the announcement with the TTY.

As an alternative to creating a TTY hunt group, you can use vectors to process TTY calls. With vectors, you can allow TTY callers and voice callers to use the same telephone number. In this case, you can also record a single announcement that contains both TTY signaling and a voice recording.

Hardware requirements for Hunt Groups

The Hunt Groups feature requires the following hardware:

- None

Administering Hunt Groups

The following steps are part of the administration process for the Hunt Groups feature:

- [Setting up hunt groups](#)
- [Changing hunt groups](#)
- [Setting up queues for hunt groups](#)
- [Adding hunt group announcements](#)
- [Setting up night service for hunt groups](#)

This section describes:

- Any prerequisites for administering the Hunt Groups feature
- The screens that are used to administer the Hunt Groups feature
- Complete administration procedures for the Hunt Groups feature

Prerequisites for administering Hunt Groups

You must complete the following actions before you can administer the Hunt Groups feature:

- None

Screens for administering Hunt Groups

Screen Name	Purpose	Fields
<i>Announcements/Audio Sources</i>	Record announcements for hunt groups.	As needed
<i>Class of Service</i>	Change COS from default of 1.	As needed
<i>Coverage Paths</i>	Set up coverage paths for hunt groups.	• Ext • Type • COR • TN • Name • Others as needed
<i>Hunt Group</i>	Add or change hunt groups.	As needed
<i>Call Center Optional Features</i>	Ensure that ACD and Expert Agent Selection (EAS) are set to y to use these call distribution methods for hunt groups.	• ACD • Expert Agent Selection (EAS)
<i>Trunk Groups</i>	Set up night service for hunt groups.	Night Service Incoming Destination

Setting up hunt groups

To set up a hunt group, you need the following information:

- The telephone number for the hunt group
- The number of people answering calls
- How calls are answered

To set up a hunt group:

- 1 Type **add hunt-group next**. Press **Enter**.

The system displays the **Hunt Group** screen ([Figure 185, Hunt Group screen](#), on page 695). The **Group Number** field is automatically filled in with the next available hunt group number. In the following example, the group number is **4**.

Figure 185: Hunt Group screen

Page 1 of X

HUNT GROUP

Group Number: 4__	ACD? _
Group Name: _____	Vector? _
Group Type: _____	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: _____	
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? _	

- 2 In the **Group Name** field, type the name of the group.
- 3 In the **Group Extension** field, type the hunt group extension number.
- 4 In the **Group Type** field, type the code for the call distribution method that you choose. For more information on Group Type codes, see [Call Distribution methods for hunt group types](#) on page 689.
- 5 Press **Enter** to save your changes.

The COS (Class of Service) for all hunt groups defaults to 1. Therefore, any changes to COS 1 on the **Class of Service** screen changes the COS for all hunt groups. A **COS** field does not appear on the **Hunt Group** screen.

- 6 Scroll through the pages of the **Hunt Groups** screen to find the **Group Member Assignments** page ([Figure 186, Hunt Group screen for group member assignments](#), on page 696).

Figure 186: Hunt Group screen for group member assignments

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)	
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 :			26 :

More Members Exist

- 7** In the **Ext** field, type the extensions of the agents that you want in the hunt group.
- 8** Press **Enter** to save your changes.

For a complete description of all fields on the *Hunt Group* screen, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Changing hunt groups

To make changes to a hunt group:

- 1** Type **change hunt-group *n***, where *n* is the number of the hunt group. Press **Enter**.
The system displays the *Hunt Group* screen (see [Figure 185, Hunt Group screen](#), on page 695).
- 2** Change the necessary fields.
- 3** Press **Enter** to save your changes.

Setting up queues for hunt groups

You can configure Communication Manager to place hunt group calls that cannot be answered immediately in a queue.

You can administer the number of calls that can wait in the queue. You can also administer the system to provide a warning when the number of calls in the queue exceeds a certain value, and when a call waits in the queue longer than a specified number of seconds.

To set up a hunt group queue:

- 1 Type **change hunt-group *n***, where *n* is the number of the hunt group to change. Press **Enter**.
The system displays the **Hunt Group** screen (see [Figure 185, Hunt Group screen](#), on page 695).
- 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.
- 4 In the **Calls Waiting Threshold** field, type the number of calls that can be in the queue before the queue status lamps flash.
- 5 In the **Time Warning Threshold** field, type the number of seconds for a call to wait in the queue before the queue status lamps flash.
- 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.

You assign the recorded announcements to an extension. You can type **display announcements** to find the extensions of recorded announcements that are already assigned to extensions. You can use the same announcement for more than one hunt group. For more information on recording announcements, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

To add an announcement to a hunt group queue:

- 1 Type **change hunt-group *n***, where *n* is the number of the hunt group to which you want to add the announcement. Press **Enter**.
The system displays the **Hunt Group** screen ([Figure 187, Hunt Group screen](#), on page 697).

Figure 187: Hunt Group screen

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: _____ Delay (sec): _____

Second Announcement Extension: _____ Delay (sec): _____ Recurring? ☐

- 2 In the **First Announcement Extension** field, type the extension of the announcement that you want callers to hear.

- 3 In the **First Announcement Delay (sec)** field, type the number of seconds you want the caller to wait before hearing the first announcement. If you set the delay announcement interval to **0**, callers automatically hear the announcement immediately. This is called a *forced first announcement*.
- 4 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. The destination you administer can be:

- An extension
- A recorded announcement extension
- A vector directory number
- Another hunt group extension
- An attendant

To administer night service for a hunt group:

- 1 Type **change hunt-group *n***, where *n* is the number of the hunt group for which you want to administer night service. Press **Enter**.

The system displays the **Hunt Group** screen ([Figure 188, Hunt Group screen](#), on page 698).

Figure 188: Hunt Group screen

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 the night service extension to which calls are routed.
- 3 Press **Enter** to save your changes.
- 4 Program a night service button, so members of the hunt group can activate and deactivate night service. For more information on how to program feature buttons, [click here](#), or see the Screen Reference section of the *Administrator's Guide for Avaya Communication Manager*.

Reports for Hunt Groups

The following reports provide information about the Hunt Groups feature:

- None

Considerations for Hunt Groups

This section provides information about how the Hunt Groups feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Hunt Groups feature under all conditions.

- Members assigned to multiple hunt groups

An extension can be a member of more than one hunt group. However, a telephone, even a multi-appearance telephone, can receive only one hunt group call at a time. On a multi-appearance telephone, all appearances must be idle to receive a hunt group call.

You can assign a Coverage Incoming Call Indicator (ICI) button to a multi-appearance telephone or an attendant console. When a member receives a call for the hunt group that is associated with the ICI button, the status lamp lights.

- Automatic Call Distribution (ACD) agents as hunt group members

Do not include agents in an ACD split in non-ACD hunt groups if the agents also receive ACD split calls. The system distributes all ACD calls to agents in a split before the system distributes hunt group calls.

When you change an ACD split to a non-ACD hunt group, each agent in the split must enter the Hunt Group Busy deactivation code to receive calls for that hunt group. If the agent has an AUX-work button, the status 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 that is 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 use Class of Restrictions (COR) to restrict an extension from receiving calls other than those calls to the hunt group to which the extension is assigned. You can also restrict extensions on Communication Manager from calling the extension of the hunt group.

- 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 telephone on Communication Manager can dial that extension to call the hunt group. If a trunk group can pass digits from the central office (CO) to Communication Manager, for example, over a DS1 trunk group, a caller can also dial a 7-digit number. The 7-digit number consists of a specified prefix and the extension of the hunt group.

If a trunk group cannot pass digits from the CO to Communication Manager, the system can connect incoming calls to a hunt group only if the trunk group has the hunt group extension as its primary destination. This requirement includes trunk groups for incoming listed directory number (LDN) calls, international exchange calls, 800 service calls, and automatic tie-trunk calls.

Interactions for Hunt Groups

This section provides information about how the Hunt Groups feature interacts with other features in the system. Use this information to ensure that you receive the maximum benefits of the Hunt Groups feature in any feature configuration.

- Attendant Call Waiting

Attendant Call Waiting does not work for calls that the attendant sends to a hunt group. Attendant Call Waiting 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)

ACD does not work with circular station hunting.

- Call Detail Recording (CDR)

For each call, the system can record the associated hunt group extension or member extension that answers.

- Internal Automatic Answer (IAA)

Internal calls to a hunt group member are eligible for Internal Automatic Answer (IAA).

- Leave Word Calling (LWC)

A hunt group can receive and store LWC messages. The following entities 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 telephone display, and proper authorization. If the message retriever is a member of the hunt group, you can assign a remote Automatic Message Waiting lamp to the retriever. The lamp indicates when the hunt group has an LWC message.

- **Night Service**
When the Night Service destination for a hunt group is another hunt group, callers hear the forced announcement of the first hunt group, if a forced first announcement is 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 three-burst ring.
- **Queuing**
Queuing does not work with circular station hunting.
- **Terminating Extension Group (TEG)**
A TEG cannot be a member of a hunt group.
- **Vectoring**
Call vectoring does not work with circular station hunting.

Individual Attendant Access

Use the Individual Attendant Access feature to allow users to call a specific attendant console.

Detailed description of Individual Attendant Access

This section provides a detailed description of the Individual Attendant Access feature.

If you have more than one attendant console, you can assign an extension to each console. Users can then dial the assigned extension to call an attendant directly.

With individual attendant extensions, you can also allow attendants to use features that an attendant group cannot use. For example, you can assign individual attendant extensions to hunt groups.

Hardware requirements for Individual Attendant Access

The Individual Attendant Access feature requires the following hardware:

- Attendant console

Administering Individual Attendant Access

The following steps are part of the administration process for the Individual Attendant Access feature:

- [Assigning an extension to an attendant console](#)

This section describes:

- Any prerequisites for administering the Individual Attendant Access feature
- The screens that you use to administer the Individual Attendant Access feature
- Complete administration procedures for the Individual Attendant Access feature

Prerequisites for administering Individual Attendant Access

You must complete the following actions before you can administer the Individual Attendant Access feature:

- None

Screens for administering Individual Attendant Access

Screen name	Purpose	Fields
<i>Attendant Console</i>	Assign an extension to an attendant console.	Extension

Assigning an extension to an attendant console

To assign an extension to an attendant console:

- 1 Type **add attendant *n***, where *n* is a number between 1 and 28 that you want to assign to the attendant console. Press **Enter**.

The system displays the *Attendant Console* screen ([Figure 189, Attendant Console screen](#), on page 704).

Figure 189: Attendant Console screen

add attendant console		Page 1 of 4		
ATTENDANT CONSOLE 2				
Type:	Name: operator			
Extension:	Group: 1	Auto Answer: none		
Console Type:	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		
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 **Extension** field, type a valid extension that conforms to your dial plan.
For information on how to set up the rest of an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.
- 3 Press **Enter** to save your changes.

Reports for Attendant Individual Attendant Access

The following reports provide information about the Individual Attendant Access feature:

- None

Considerations for Individual Attendant Access

This section provides information about how the Individual Attendant Access feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Individual Attendant Access under all conditions. The following considerations apply to Individual Attendant Access:

- None

Interactions for Individual Attendant Access

This section provides information about how the Individual Attendant Access feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Individual Attendant Access in any feature configuration.

- None

Individual Attendant Access

Interactions for Individual Attendant Access

Inter-PBX Attendant Service

Use the Inter-PBX Attendant Service (IAS) feature to allow attendants who support multiple locations, to work at a single location.

Detailed description of Inter-PBX Attendant Service

This section provides a detailed description of the Inter-PBX Attendant Service (IAS) feature.

With IAS, attendants can work at one location while the attendants support users at other locations. The system routes any incoming trunk calls that are to a user location, and any attendant-seeking calls, over tie trunks to the attendant location.

Hardware requirements for Inter-PBX Attendant Service

The Inter-PBX Attendant Service (IAS) feature requires the following hardware:

- An attendant console

Administering Inter-PBX Attendant Service

This section contains prerequisites and the screens for administering the Inter-PBX Attendant Service (IAS) feature.

The following steps are part of the administration process for the Inter-PBX Attendant Service feature:

- [Enabling Inter-PBX Attendant Service](#)

This section describes:

- Any prerequisites for administering the Inter-PBX Attendant Service
- The screens that you use to administer the Inter-PBX Attendant Service
- Complete administration procedures for the Inter-PBX Attendant Service

Prerequisites for administering Inter-PBX Attendant Service

You must complete the following actions before you can administer the Inter-PBX Attendant Service feature:

- Set up an attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.
- View the **Optional Features** screen, and ensure that the Centralized Attendant field is set to **n**. If the Centralized Attendant field is set to **y**, your system is not enabled for the Inter-PBX Attendant Service feature. Contact your Avaya representative before you continue with this procedure.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

Screens for administering Inter-PBX Attendant Service

Screen name	Purpose	Fields
<i>Console Parameters</i>	Enable IAS.	• IAS Att. Access Code • IAS (Branch)? • IAS Tie Trunk Group No.

Enabling Inter-PBX Attendant Service

To enable Inter-PBX Attendant Service:

- 1** Type **change console-parameters**. Press **Enter**.

The system displays the **Console Parameters** screen ([Figure 190, Console Parameters](#), on page 709).

Figure 190: Console Parameters

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	
IAS (Branch)? y	Night Service Act. Ext.:
IAS Att. Access Code:	IAS Tie Trunk Group No.:
Backup Alerting? n	Alternate FRL Station:
Attendant Vectoring VDN:	DID-LDN Only to LDN Night Ext? n

- 2 In the IAS Att. Access Code field, type the extension of the attendant group at the main server that runs Communication Manager.
You must type **y** in this field, if the IAS (Branch)? field is set to **y**.
The system displays the IAS Att. Access Code field only if the Centralized Attendant field on the **Optional Features** screen is set to **n**.
- 3 In the IAS (Branch)? field, type **y**.
The system displays this field only if the Centralized Attendant field on the **Optional Features** screen is set to **n**.
- 4 In the IAS Tie Trunk Group No. field, type the number of the tie trunk group to the main IAS location. Valid entries for the DEFINITY R, CSE, and SI are **1** through **666**. Valid entries for the S8300 Media Server, S8700 IP-Connect, and S8700 Multi-Connect are **1** through **2000**.
You must complete this field if the IAS (Branch)? field is set to **y**.
The system displays this field only if the Centralized Attendant field on the **Optional Features** screen is set to **n**.
- 5 Press **Enter** to save your changes.

Reports for Inter-PBX Attendant Service

The following reports provide information about the Inter-PBX Attendant Service (IAS) feature:

- None

Considerations for Inter-PBX Attendant Service

This section provides information about how the Inter-PBX Attendant Service (IAS) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Inter-PBX Attendant Service under all conditions. The following considerations apply to Inter-PBX Attendant Service:

- None

Interactions for Inter-PBX Attendant Service

This section provides information about how the Inter-PBX Attendant Service (IAS) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Inter-PBX Attendant Service in any feature configuration.

- Centralized Attendant Service

IAS does not work with Centralized Attendant Service

Intercom

Use the Intercom feature to administer a button that calls a predefined extension when the button is pressed. You can also use Intercom feature to allow one user to call another user in a predefined group just by pressing a couple of buttons.

Intercom supports the following capabilities:

- Automatic Intercom
- Dial Intercom

Detailed description of Intercom

This section provides a detailed description of the Intercom feature.

To control which telephones can make intercom calls to each other, you put the telephones in groups called “intercom groups.” Once you add a set of telephones to the group, users can make intercom calls by administering one or both of the following feature buttons on their telephones:

- Automatic Intercom
Users use this button to call one predefined telephone in the same intercom group. You specify the destination extension for this button.
- Dial Intercom
Users in an intercom group use this button to call anyone else in the same group. The user lifts the handset, presses the Dial Intercom button, and then dials a 1digit or a 2 digit code for the extension.

Telephones with both of these capabilities can belong to the same intercom group.

Intercom groups

- You can create up to 32 intercom groups on one server that runs Avaya Communication Manager.
- Each group can contain up to 32 extensions in it.
- You can assign the same extension to different groups.
- Intercom calls are possible only 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.

Telephones

- You can assign any type of telephone to an intercom group. However, only multiappearance telephones can make and receive intercom calls. Single-line telephones can only receive intercom calls. Multiappearance telephones must have at least one open or available call appearance to receive intercom calls.
- An intercom call makes a unique alerting sound. If the telephone has an intercom button with a status lamp, the lamp also flashes.
- An automatic intercom connection between two telephones, even if the Class of Restriction (COR) does not allow other calls between them.

Hardware requirements for Intercom

The Intercom feature requires the following hardware:

- None

Administering Intercom

The following steps are part of the administration process for the Intercom feature:

This section describes the screens that you use to administer the Intercom feature.

Screens for administering Intercom

Screen name	Purpose	Fields
<i>Intercom Group</i>	Create groups for intercom.	• DC • Ext • Group Number

Reports for Intercom

The following reports provide information about the Intercom feature:

- None

Considerations for Intercom

This section provides information about how the Intercom feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Intercom under all conditions. The following considerations apply to Intercom:

- None

Interactions for Intercom

This section provides information about how the Intercom feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Intercom in any feature configuration.

- Bridged Appearances
Bridged appearances cannot 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 to a destination that is off the network.
- Call Pickup and 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.

Internal Automatic Answer

Use the Internal Automatic Answer (IAA) feature to provide a convenient, hands-free way to answer internal calls to users who have multifunction stations with a speakerphone or a headphone.

Detailed description of Internal Automatic Answer

This section provides a detailed description of the Internal Automatic Answer (IAA) feature.

With IAA, a user can answer an eligible call automatically if the user activates IAA at the answering telephone, and the telephone can accept an incoming call. A telephone cannot accept an incoming call if the telephone is off-hook, in the process of dialing digits, or has a call on hold.

The following internal calls are eligible for IAA: You administer the ***Feature-Related System Parameters*** screen:

- Station-to-station voice calls, with both telephones on the same server. These calls include redirected intraswitch calls. To use IAA for these calls, you must set the **Internal Auto-Answer of Attd-Extended/Transferred Calls** field to **transferred** or **both**.
- Internal call from another node in a Distributed Communications System (DCS) configuration. These calls are from an internal, non attendant telephone on that node and includes redirected inter-DCS calls. To use IAA for these calls, you must set the **Internal Auto-Answer of Attd-Extended/Transferred Calls** field to **transferred** or **both**.
- Attendant-extended external calls. You must set the **Internal Auto-Answer of Attd-Extended/Transferred Calls** field to **attd-extended** or **both**.

The following calls are ineligible for IAA:

- Calls from public-network trunks, including Private 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 an overflow of Emergency Access to the Attendant calls in the queue.
- Calls when the **Active Station Ringing** field of the receiving telephone is set to **continuous** on the ***Features-Related System Parameters*** screen.

Feature operations

With IAA, you can assign a single programmable feature button to telephones. When the user presses the IAA feature button, the button lamp lights, and the system activates IAA. When the user presses the same button again, the system deactivates IAA, and turns off the status lamp. Pressing the feature button has no effect on a an active call or a ringing call. The IAA button can 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.

When IAA answers a call, the calling telephone receives a tone. The called telephone answered automatically by a telephone with IAA. The called telephone receives a tone in the form of a ring ping. The called phone then goes off-hook to automatically answer an IAA-eligible call. The system turns on the answering telephone's speaker and the microphone of the called phone.

If a user has IAA active and is currently busy on a call, or in the process of dialing digits, subsequent incoming calls are treated as if IAA were not activated.

Hardware requirements for Internal Automatic Answer

The Internal Automatic Answer (IAA) feature requires the following hardware:

- None

Administering Internal Automatic Answer

This section describes the screens that you use to administer the Internal Automatic Answer (IAA) feature.

Screens for administering Internal Automatic Answer

Screen name	Purpose	Fields
<i>Feature-Related System Parameters</i>	Set up IAA at a system level.	Internal Auto-Answer of Attd-Extended/Transferred Calls

Reports for Internal Automatic Answer

The following reports provide information about the Internal Automatic Answer (IAA) feature:

- None

Considerations for Internal Automatic Answer

This section provides information about how the Internal Automatic Answer (IAA) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Internal Automatic Answer under all conditions. The following considerations apply to Internal Automatic Answer:

- Users must always deactivate IAA when the users leave the work area. If users do not deactivate IAA, the unattended station might unintentionally answer incoming calls, instead of sending all the calls.
- A 602A terminal is off-hook when the headset or the speakerphone is connected. Therefore, IAA answers a call if all other call appearances are idle.

Interactions for Internal Automatic Answer

This section provides information about how the Internal Automatic Answer (IAA) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Internal Automatic Answer in any feature configuration.

- Attendant Console
IAA is unavailable with Attendant Console.
- Automatic Answer
You cannot administer both IAA and Automatic Answer simultaneously on the same telephone.
- Automatic Call Distribution (ACD)
Calls that are directed to an ACD split are eligible for IAA.
- Automatic Callback
Callback calls by way of Automatic Callback are not answered automatically by IAA.
- Automatic Circuit Assurance (ACA)
Calls that are generated by ACA are ineligible for IAA.
- Bridged Call Appearance - Multiappearance Telephone
Calls that terminate on a bridged call appearance are ineligible for IAA at the bridged station, even if the bridged station has IAA active. However, IAA can be used by the principal station to answer the call.
- Bridged Call Appearance - Single-Line Telephone
Calls that terminate to a bridged call appearance are ineligible 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, 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 Send All Calls active.
 - The calling telephone selects Go to Cover before placing the call

Calls that are directed to a coverage answering group are ineligible for IAA.

NOTE:

If you set the coverage path for a telephone to All Calls, and that telephone activates IAA, the first coverage point hears a ring. Then the principal station automatically answers, and the coverage-simulated bridge is dropped. The coverage station rings, but is unable to answer the call, because the coverage-simulated bridge was dropped.

- **Call Forwarding**

Calls to a telephone with IAA and Call Forwarding active are forwarded, and are unanswered by the station dialed.

NOTE:

If the forwarded-to telephone is internal and has IAA active, the forwarded-to telephone automatically answers the redirected call.

- **Call Park**

If you are using Deluxe Paging and Call Park times out, the call returns to the originating telephone that parked the call, and is eligible for IAA.

- **Call Pickup**

IAA can answer internal calls to a telephone in a Call Pickup group. 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**

IAA can answer internal conference calls. If more than one party has joins a conference call through automatic answer, the parties remain connected until the parties disconnect, or the controlling party drops the call.

- **Data Call Setup**

Data calls are not eligible for IAA.

- **Direct Department Calling (DDC) and Uniform Call Distribution (UCD)**

Internal calls to a member of a DDC or a member of a UCD group member are eligible for IAA.

- **Distributed Communications System (DCS)**

If a call is from an internal telephone on another server or switch in a DCS configuration, 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 selects Go to Cover before placing a call.

- **ISDN-BRI**

IAA is unavailable with ISDN-BRI telephones.

- **Loudspeaker Paging - Deluxe Paging**

If Call Park times out when you are using Deluxe Paging, the call returns to the originating telephone that parked the call. Such calls are eligible for IAA.

- Ringback Queuing
Automatic calls that are generated by Ringback Queuing are ineligible for IAA.
- Send All Calls
IAA does not apply to calls to extensions with Send All Calls is active.
- Terminating Extension Group (TEG)
Calls to a TEG extension are ineligible for IAA. However, calls to an individual extension are eligible.

ISDN Service

Use the Integrated Services Digital Network (ISDN) Service feature to provide a message-oriented signaling method that allows information to be sent along with a call. The ISDN Service feature also 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 Communication Manager can be connected to an ISDN by way of the standard frame formats: Basic Rate Interface (BRI) and the Primary Rate Interface (PRI).

Detailed description of ISDN Service

This section provides a detailed description of the Integrated Services Digital Network (ISDN) Service feature.

The ISDN Service feature 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 ISDN 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 feature description 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 help you in planning, installing, and administering ISDN trunks.

ISDN supports the following features:

- 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), but only with ISDN-PRI.
- Generalized Route Selection (GRS)
- Call Identification Display - Calling Party Number (CPN) and Billing Number (BN)
- Administered Connections and Access Endpoints
- Interworking, or the mixture of ISDN and non-ISDN trunks and stations
- Wideband Switching (H0, H11, H12, and NxDS0, but only with ISDN-PRI.
- QSIG Multivendor Connectivity
- Lookahead Interflow
- Lookahead Routing
- Usage Allocation

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 Communication Manager offers several administrable protocols, each of which provides a different set of ISDN services. 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, Communication Manager interfaces with a wide range of other products including servers, network switches, and host computers. These products include earlier releases of servers running Communication Manager, public network switches (for example, AT&T 4ESS, Lucent 5ESS, and Northern Telecom DMS250), and other products adhering to the ISDN signaling protocol.

[Figure 191, ISDN network configuration](#), on page 722 shows an example of how ISDN is used in private and public-network configurations. For example, ISDN can be used to connect a switch to a public-switched network, to other switches, and to computers:

Figure 191: ISDN network configuration

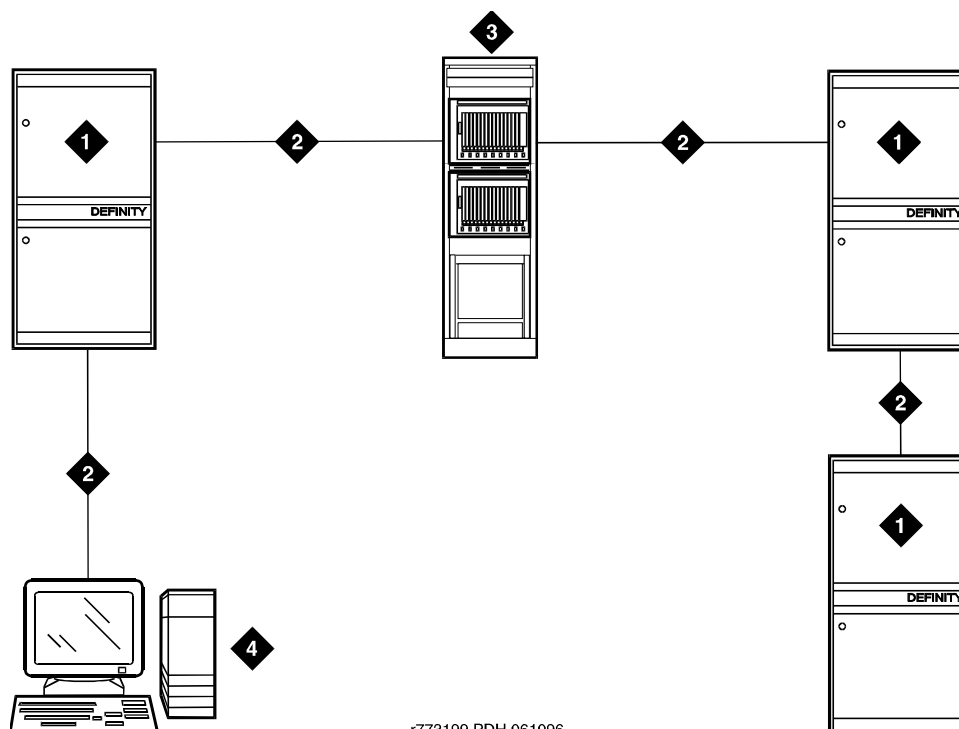


Figure notes

1	Avaya Media Server	3	Public switched network
2	ISDN trunk	4	Host computer

AT&T Switched Network protocol

Communication Manager 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.

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, either the DCS or ISDN call identification information can be displayed. 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.

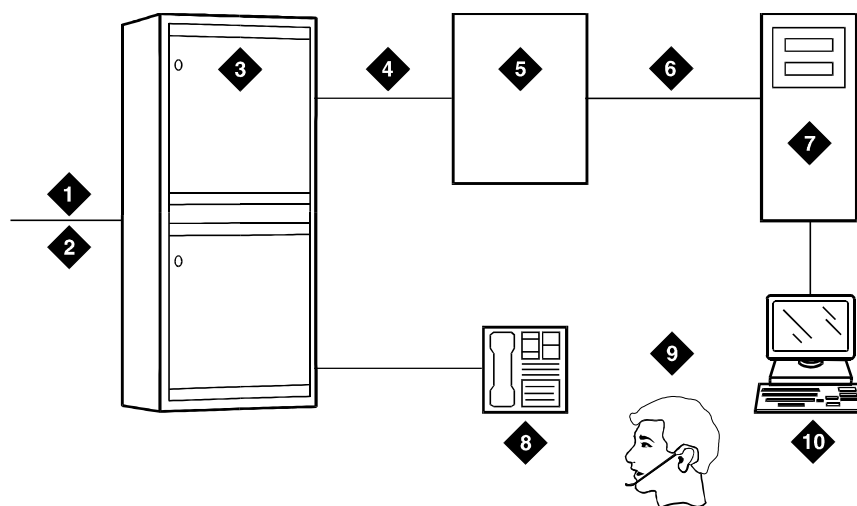
CPN/BN to Host Call Identification

The CPN/BN to Host Call Identification enables CPN and BN information to be passed from Communication Manager 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 additional Communication Manager 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.

[Figure 192, CPN- and BN-to-host configuration](#), on page 724 shows a simplified diagram of a CPN- and BN-to-host arrangement. The ISDN Gateway is a UNIX or MS-DOS computer connected to Communication Manager on one side and to a host computer on the other side. The media server connection is over a synchronous interface with BX.25 protocol.

Figure 192: CPN- and BN-to-host configuration



781861 C.JL 060996

Figure notes

1	ISDN trunk	6	Existing interface
2	SID/ANI	7	Host computer
3	Avaya Media Server	8	Telephone
4	BX.25	9	ACD agent position
5	ISDN Gateway	10	Data terminal

Private network services

In addition to providing access to switched-public networks, ISDN provides private-network services by connecting Avaya Communication Manager in an ETN, 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 servers or switches. In this case, the main server or 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 server or 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 server or 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 media servers or 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 Communication Manager 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 by way of 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 by way of 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 Communication Manager 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, but with ISDN-PRI only
- Wideband Switching, but with ISDN-PRI only
- Call-by-Call Service Selection

Calling Line Identification

Calling Line Identification for NI-2 is essentially CPN identification, as previously described.

Non-Facility Associated Signaling (ISDN-PRI only)

NEAS 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 facility that contains the D-channel.

D-Channel Backup (ISDN-PRI only)

D-Channel Backup is provided to improve reliability in the event of a signaling-link failure.

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.

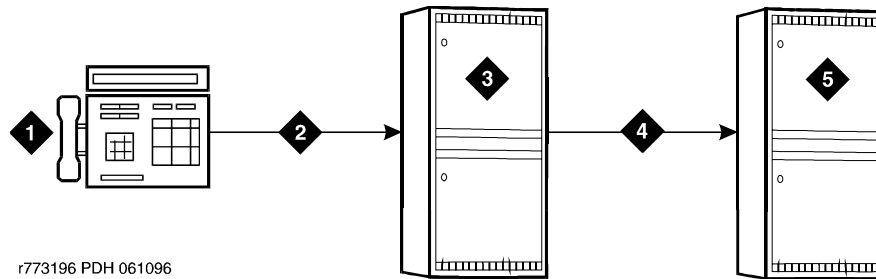
ISDN interworking

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).

Communication Manager supports 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. [Figure 193, ISDN and non-ISDN interworking](#), on page 727 shows an example of interworking.

Figure 193: ISDN and non-ISDN interworking



r773196 PDH 061096

Figure notes

1	Call from network to Switch B	4	Non-ISDN trunk
2	ISDN trunk	5	Switch B
3	Switch A		

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.

Communication Manager 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 by way of Equal Access or 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 a 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 a 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 a 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 a 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 a DEFINITY server, the (public or private) ISDN network may provide the connected party's name to the Avaya Communication Manager, 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 a 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 media servers or switches that are 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 server running Communication Manager.
- 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 shown by another server. 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 Terminating Extension Group (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 (FRL) and Traveling Class Mark (TCM)

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 modifies ISDN messaging operations in systems that use TGE/TGU trunks to network satellite servers or switches. Messaging from Communication Manager provides appropriate ringback or busy tone to the calling party.

Hardware requirements for ISDN Service

The ISDN Service feature requires the following hardware:

- None

Administering ISDN Service

This section describes the screens that you use to administer the ISDN Service feature.

Screens for administering ISDN Service

Screen name	Purpose	Fields
<i>Access Endpoint</i>	Administer ISDN Service.	- Access Endpoints - Wideband Access Endpoint
<i>Trunk Groups</i>	Administer incoming calls and trunk groups usage and allocation.	All
<i>ISDN Numbering - Private</i>	Administer private numbering plans.	All
<i>ISDN Numbering - Public/Unknown</i>	Administer ISDN call identification displays.	All

Reports for ISDN Service

The following reports provide information about the ISDN Service feature:

- None

Considerations for ISDN Service

This section provides information about how the ISDN Service feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of ISDN Service under all conditions. The following considerations apply to ISDN Service:

- None

Interactions for ISDN Service

This section provides information about how the ISDN Service feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of ISDN Service in any feature configuration.

- **Australia Malicious Call Trace (MCT)**
Avaya Communication Manager with a BRI connection to the Australian public network can notify the network if a user in a private network invokes the MCT feature. This service works on Australian national connections.
- **Direct Inward Dialing (DID)**
Some public network operators may not offer full DID service on BRI trunks, but instead may offer the BRI equivalent, which typically is called 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. 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 a DEFINITY server, DCS displays generally override ISDN displays. However, with Avaya Communication Manager, 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 Communication Manager 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 Communication Manager does not support wideband switching on BRI connections.

Last Number Dialed

Use the Last Number Dialed feature to automatically redial the last telephone number that was dialed from the telephone, or from a bridged appearance of the telephone.

Detailed description of Last Number Dialed

This section provides a detailed description of the Last Number Dialed feature.

With the Last Number Dialed feature, users can to automatically redial the last telephone number that was dialed from the telephone, or from a bridged appearance of the telephone.

The system saves the first 24 digits of the last telephone number that was dialed. The system saves the digits for calls that the user places with either manual dialing or Abbreviated Dialing. When a user presses the Last Number Dialed button, or dials the feature access code (FAC) for Last Number Dialed, the system places the call again.

Hardware requirements for Last Number Dialed

The Last Number Dialed feature requires the following hardware:

- None

Administering Last Number Dialed

This section provides the screens that you need to administer the Last Number Dialed feature:

Screens for administering Last Number Dialed

Screen name	Purpose	Fields
<i>Attendant Console</i>	Assign a Last Number Dialed feature button to an attendant console.	Feature Button Assignments • last-numb
<i>Feature Access Code (FAC)</i>	Assign an FAC for Last Number Dialed.	Last Number Dialed Access Code
<i>Station - multiappearance</i>	Assign a Last Number Dialed feature button for a user.	Button/Feature Button Assignments • last-numb

Reports for Last Number Dialed

The following reports provide information about the Last Number Dialed feature:

- None

Considerations for Last Number Dialed

This section provides information about how the Last Number Dialed feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Last Number Dialed under all conditions. The following considerations apply to Last Number Dialed:

- When the user presses the last number dialed button, the system outputs any special characters that are stored in the Abbreviated Dialing button that was used to place the previous call. Such characters include Pause, Wait, Mark, and Suppress.
- Any delays that a user encountered when the user dialed the call manually, are not repeated when the user uses the Last Number Dialed feature.
- The system does not save Last Number Dialed information to disk, tape, or flash card. The system never saves manually dialed end-to-end, signaling digits.
- A user can enter a partial number, hang up, and use Last Number Dialed, and manually enter the remaining digits. If the user calls from a display telephone, the system does not display the digits that the user enters manually. However, the system completes the call.

Interactions for Last Number Dialed

This section provides information about how the Last Number Dialed feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Last Number Dialed in any feature configuration.

- Abbreviated Dialing
If the previously called number is in an Abbreviated Dialing privileged list, and the COR of the user prevents the user from dialing the number, the system uses Intercept Treatment when the user presses Last Number Dialed. To redial the number, the user must again use the Abbreviated Dialing privileged list.
- Automatic Callback
Users can use Automatic Callback after the users use Last Number Dialed on a call to an internal voice terminal.
- Bridged Call Appearance
Last Number Dialed causes the last number that was dialed from the telephone, or a bridged appearance of the telephone, to be redialed.
- Centralized Attendant Service (CAS)
If a CAS attendant attempts to extend a call with Last Number Dialed, the system does not complete the call.

Leave Word Calling

Use the Leave Word Calling (LWC) feature to allow internal system users to leave a short pre-programmed message for other internal users. When the message is stored, the Automatic Message Waiting lamp lights on the called telephone. Users can retrieve LWC messages on a telephone display, Voice Messaging Retrieval, or the AUDIX system. Messages can be retrieved in English, French, Italian, Spanish, or a user-defined language.

Detailed description of Leave Word Calling

This section provides a detailed description of the Leave Word Calling (LWC) feature.

LWC electronically stores a standard message. For example:

CARTER, ANN 2/7 10:45a 2 CALL 3124

This message means that Ann Carter called two times, the last time on the morning of February 7 at 10:45 a.m. She wants a return call to extension 3124.

When the system receives identical messages, the system updates only the date and the time, and number of messages. If nine or more identical messages accumulate, the count remains at nine, and the system updates only the date and time.

Messages can be stored by calling users, called users, and covering users as follows:

- Calling user
 - 4** Press the LWC button or dial the LWC access code.
 - 5** Dial the desired number.
 - 6** Before the call is answered, if you are a multiappearance voice-terminal user, press LWC. If you are a single-line voice-terminal user, press Recall and dial the access code.
 - 7** After the call is answered, press LWC or Recall and dial the access code.
- Called user
 - 1** The user answers the call.
 - 2** The called user presses LWC to leave a message for the calling user to return the call.
 - 3** A called user can store an LWC message by dialing the LWC access code only if the called user has an analog voice terminal.
- Covering user

A covering user can be through Call Coverage, Call Pickup, or Call Forwarding All Calls.

 - 1** The covering user answers the call.
 - 2** The covering user presses Coverage Callback to store a message for the called user that tells them to return a call the calling user.
 - 3** After answering the call, the covering user presses LWC to leave a call-me message for the originally called user.

In addition, a user that was placed on hold can activate LWC and leave a message for the holding user to place a return call.

A caller who leaves a LWC message can cancel that message, if the message was not already retrieved. To cancel the message, the calling user lifts the handset, presses LWC Cancel or dials the access code, and then dials the extension of the called party.

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 voice terminal. The Remote Automatic Message Waiting lamp is a status lamp associated with a button assigned for this purpose. Thus, the lamp on an assistant's telephone can light when an executive receives a LWC message. If the executive calls to retrieve messages, the assistant knows at a glance if any messages were left.

Users without Voice Terminal 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 capability restricts a voice terminal and the Unlock function releases the restriction. To activate Lock, the users dial 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 voice terminal. These functions apply only to the voice terminal where the function is active. You can assign a status lamp to show the lock status of the voice terminal.

Voice synthesis

Avaya solutions can also provide voice synthesis for LWC, depending on which voice-synthesis circuit pack is installed. This voice synthesis can be either in English or Italian.

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.

With the Leave Word Calling Log External Calls capability, the system can monitor when an external call is not answered. The server keeps a record of as many as 15 calls, provided that the caller identification is available, and the message lamp on the telephone lights. The displays on the telephone shows the names and numbers of unsuccessful callers.

Hardware requirements for Leave Word Calling

The Leave Word Calling feature requires the following hardware:

- None

Administering Leave Word Calling

The following steps are part of the administration process for Leave Word Calling:

- None

Considerations for Leave Word Calling

This section provides information about how the Leave Word Calling (LWC) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Leave Word Calling feature under all conditions. The following considerations apply to the Leave Word Calling feature:

- 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% of capacity, the status lamps that are associated with all Coverage Message Retrieval buttons in the system flash. These lamps continue to flash until the stored-message level falls below 85%. Authorized retrievers can selectively delete messages to gain storage space. The system does not automatically purge old messages.
- LWC messages cannot be stored, canceled, or retrieved for Vector Directory Number extensions.

Interactions for Leave Word Calling

This section provides information about how the Leave Word Calling (LWC) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Leave Word Calling in any feature configuration.

- **AUDIX Interface**
LWC Cancel cannot be used to cancel an AUDIX message.
- **Bridged Call Appearance**
A LWC message that is 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. LWC provides the Coverage Callback option. Also, a caller can activate LWC for the called party even if the call was answered by a covering user.
- **Centralized Attendant Service (CAS)**
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 is activated for a party on a conference or transfer, the origination of the conference or the transfer 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.
- **Distributed Communications System (DCS)**
LWC works with DCS, but only for 4- digit and 5-digit extension dial plans. LWC works with QSIG for all dial plans of 3- through 7-digits.

- Expert Agent Selection (EAS)

When an EAS agent is logged into a telephone, the agent can only retrieve LWC messages that are left for the login ID of the agent. To retrieve LWC messages that are left for that telephone, the agent must log out.

When an EAS agent is logged into a telephone, the Message lamp of the telephone 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 login ID of the agent.

- Message Waiting Indicator (MWI)

As QSIG does not specify a standard way to light the MWI lamp upon receipt of an LWC message, Avaya's implementation requires systems running Avaya Communication Manager in order to work properly.

- Vector Directory Number (VDN)

LWC messages cannot be stored, cancelled, or retrieved through VDN extensions.

Line Lockout

Use the Line Lockout feature to remove a user with a single-line telephone from service when the user does not hang up after the user receives dial tone or intercept tone.

Detailed description of Line Lockout

This section provides a detailed description of the Line Lockout feature.

Lockout occurs when:

- A user does not hang up after the other party on a call disconnects.
 - The user receives the dial tone for 10 seconds, and then receives the intercept tone for the interval that you administer.
 - You can administer the system to play a special howler tone before the system takes the telephone out of service.
 - If the handset remains off-hook, the system takes the telephone out of service.
- A user pauses for ten seconds between digits when the user dials numbers.
 - The user receives intercept tone for 30 seconds.
 - If the handset remains off-hook, the system takes the telephone out of service.

The telephone remains out of service until the user hangs up.

Hardware requirements for Line Lockout

The Line Lockout feature requires the following hardware:

- None

Administering Line Lockout

This section describes the screens that you use to administer Line Lockout.

Screens for administering Line Lockout

Screen name	Purpose	Fields
<i>Feature-Related System Parameters</i>	Specify the number of seconds that the system waits before the system removes the user telephone from service, after the system generates the warning tone.	Line Intercept Tone Timer
	Specify the tone that the system generates for the last user on a call, until the user hangs up or the system generates the tone for 45 seconds.	Station Tone Forward Disconnect
	Enter the number of seconds that the system allows a telephone with an Off-Hook Class of Service (COS) to remain off-hook before the system sends an emergency call to the attendant.	Time Before Off-hook Alert
<i>System Parameters Country-Options</i>	Enable the system to disconnect calls that are unanswered	Disconnect on No Answer by Call Type
	Enable the system to generate howler tone for users, before the system removes the user telephone from service.	Howler Tone After Busy

Reports for Line Lockout

The following reports provide information about the Line Lockout feature:

- None

Considerations for Line Lockout

This section provides information about how the Line Lockout feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Line Lockout under all conditions. The following considerations apply to Line Lockout:

- The out-of-service condition that Line Lockout provides does not tie up switching facilities.
- Line Lockout does not apply to multiappearance telephones.

Interactions for Line Lockout

This section provides information about how the Line Lockout feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Line Lockout in any feature configuration.

- None

Listed Directory Number

Use the Listed Directory Numbers (LDN) feature to allow outside callers to access your attendant group.

Listed Directory Number supports the following capabilities:

- Attendant group access through incoming direct inward dialing (DID) trunks
- Attendant group access through incoming central office (CO) and foreign exchange (FX) trunks

Detailed description of Listed Directory Number

This section provides a detailed description of the Listed Directory Number (LDN) feature.

The system routes both incoming direct inward dialing DID calls and incoming foreign exchange FX and central office (CO) calls to an attendant group, based on how you administer the trunks.

Incoming DID trunk calls to the attendant group

Without the LDN feature, the system routes Incoming DID calls 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 DID calls to an attendant group.

Incoming FX and CO trunk calls to the attendant group

Incoming FX and CO trunks calls can terminate at an attendant group, although you can 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 entities:

- An attendant group
If you decide to terminate the call at an attendant group, the system processes the call as an LDN call.
- An extension
The extension can be a vector directory number (VDN), an Automatic Call Distribution (ACD) split, a Direct Department Calling (DDC) group, a Uniform Call Distribution (UCD) group, a remote access extension, or any system extension.

Hardware requirements for Listed Directory Number

The Listed Directory Number (LDN) feature requires the following hardware:

- None

Administering Listed Directory Number

The following steps are part of the administration process for the Listed Directory Number (LDN) feature:

- [Assigning listed directory numbers](#)
- [Assigning an incoming destination to a trunk](#)

This section describes:

- Any prerequisites for administering the Listed Directory Number feature
- The screens that you use to administer the Listed Directory Number feature
- Complete administration procedures for the Listed Directory Number feature

Prerequisites for administering Listed Directory Number

You must complete the following actions before you can administer the Listed Directory Number feature:

- None

Screens for administering Listed Directory Number

Screen name	Purpose	Fields
<i>Listed Directory Numbers</i>	Assign listed directory numbers and an optional, night-service destination.	All
• <i>CO Trunk Group</i> • <i>FX Trunk Group</i>	Assign an incoming destination to a trunk.	Incoming Destination

Assigning listed directory numbers

To assign a listed directory number:

- 1 Type **change listed-directory-number**. Press **Enter**.

The system displays the *Listed Directory Numbers* screen ([Figure 194, Listed Directory Numbers screen](#), on page 747).

Figure 194: Listed Directory Numbers screen

change listed-directory-number			Page 1 of 2
LISTED DIRECTORY NUMBERS			
Night Destination:			
Ext	Name	TN	
1:		1	
2:		1	
3:		1	
4:		1	
5:		1	
6:		1	
7:		1	
8:		1	

- 2 In the **Night Destination** field, type a one-digit to eight-digit night destination extension. The extension must be comprised of the numbers **0** through **9**. The night destination extension receives the calls to the extensions that you type in the **Ext** field, when the Night Service feature is active.

For DEFINITY® CSI, SI, S8700 IP-Connect, you can type a night service extension, a recorded announcement extension, a Vector Directory Number (VDN), an individual attendant extension, or a hunt group extension.
- 3 In the **Ext** field, type the number of the extension that you want to use for the LDN feature. You can type a one-digit to eight-digit extension number.
- 4 In the **Name** field, type the name that you use to identify the listed directory number. You can type 1 to 27 alphanumeric characters.
- 5 In the **TN** field, type the Tenant Partition number.
- 6 Repeat steps 3 through 5 for each extension that you want to use for the LDN feature.
- 7 Press **Enter** to save your changes.

Assigning an incoming destination to a trunk

To assign an incoming destination to a trunk:

- 1 Type **change trunk-group *n***, where ***n*** is the number of the trunk group to which you want to assign an incoming destination.

The system displays the **Trunk Group** screen ([Figure 195, Trunk Group screen](#), on page 748).

Figure 195: Trunk Group screen

add trunk-group next
Page 1 of x

TRUNK GROUP

Group Number: ____	Group Type: tie____	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? _	
Incoming Calling Number - Delete: ____	Insert: ____ Format: ____
Bit Rate: ____	Synchronization: ____ Duplex: ____
Disconnect Supervision - In? _ Out? _	
Answer Supervision Timeout: ____	Receive Answer Supervision? _

2 In the **Incoming Destination** field, perform one of the following actions:

- Leave the field blank if the **Trunk Type** field on the **Trunk Group** screen is not set to **auto**.
- Type the extension for the incoming calls.
- You can type any extension. However, the extension is usually for a VDN, a voice response unit, or a voice messaging system. The **Night Service** feature overrides the extension that you type in the **Incoming Destination** field.



CAUTION:

When you assign a Multi-Location Dial Plan shortened extension in a field that is designed for announcement extensions, certain administration end validations that are usually performed on announcement extensions are not performed, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Ensure that you administer the correct type of announcement for the application when you assign shortened extensions.

- Type **attd** if you want the system to route the calls to the attendant.

The system records the calls as LDN calls on the call detail recording records.

The system displays the **Incoming Destination** field, when the **Direction** field on the **Trunk Group** screen is set to **incoming** or **two-way**.

Use the **Incoming Destination** field to set the destination for all incoming calls on trunk groups such as central office (CO), foreign exchange (FX), and Wide Area Telecommunications Service (WATS), that must terminate at a single destination. The destination that you type in the **Incoming Destination** field is also the default night service destination, unless you enter a different destination in the **Night Service** field.

3 Press **Enter** to save your changes.

Reports for Listed Directory Number

The following reports provide information about the Listed Directory Number feature:

- None

Considerations for Listed Directory Number

This section provides information about how the Listed Directory Number (LDN) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Listed Directory Number under all conditions. The following considerations apply to Listed Directory Number:

- The number of listed directory numbers that you can assign depends on the configuration of your system.

Interactions for Listed Directory Number

This section provides information about how the Listed Directory Number (LDN) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Listed Directory Number in any feature configuration.

- Night Service

If you activate the Night Service capability, and a night console is not assigned or is not operational, the system routes incoming LDN calls as follows:

- Direct Inward Dialing (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 central office (CO) or foreign exchange (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.

Loss Plans

Use the Loss Plans feature to 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.

Detailed description of Loss Plans

This section provides a detailed description of the Loss Plans feature.



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. Avaya recommends that you seek technical assistance from Avaya before making any modifications to the loss plan.

The **2 Party Loss Plan** page of the **Location Parameters** 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 **Tone Generation** screens. This allows you to use different loss plans for different types of phones or different trunk groups.

The **Tone Loss Plans** page of the **Location Parameters** screen allows you to set the tone gain or loss levels (in dB) on a conference call, as well as the total gain or loss in a conference based on the number of parties.

NOTE:

The end-to-end total loss for multi-party conference calls that is administered on the **Location Parameters** screen is not always applied to a specific call. The loss applied to, for example, a 3-party conference call is calculated by adding the fixed pairwise loss for each pair of ports to the value for 2-party loss shown on the **Location Parameters** screen. If this total is *less* than the end-to-end total loss value configured for a 3-party conference, calculate the difference, and divide the difference by 2. Add 1 to this figure, and the result is the amount of loss applied to the call.

IP endpoints connected using hairpinning or direct IP-IP are not under the control of the administrable loss plan.

Hardware requirements for Loss Plans

The Loss Plans feature requires the following hardware:

- None

Administering Loss Plans

The following steps are part of the administration process for the Loss Plans feature:

This section describes:

- Any prerequisites for administering the Loss Plans feature
- The screens that you use to administer the Loss Plans feature
- Complete administration procedures for the Loss Plans feature

Prerequisites for administering Loss Plans

You must complete the following actions before you can administer the Loss Plans feature:

- Ensure that the **Digital Loss Plan Modification** field on the **Optional Features** screen is set to **y** this screen. If the field is not set to **y**, your system is not enabled for the Loss Plans feature. Contact your Avaya representative before you continue with this procedure.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

- Ensure that the **Customize** field on the **Location Parameters** screen is set to **y**.

To view the **Location Parameters** screen, type **display location-parameters**. Press **Enter**.

Screens for administering Loss Plans

Screen name	Purpose	Fields
Optional Features	Enable the Loss Plans feature.	Digital Loss Plan Modification
Location Parameters	Customize the Loss Plans feature.	Customize

Reports for Loss Plans

The following reports provide information about the Loss Plans feature:

- None

Considerations for Loss Plans

This section provides information about how the Loss Plans feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Loss Plans under all conditions. The following considerations apply to Loss Plans:

- None

Interactions for Loss Plans

This section provides information about how the Loss Plans feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Loss Plans in any feature configuration.

- None

Loudspeaker Paging

Use the Loudspeaker Paging feature to connect Avaya Communication Manager to loudspeaker systems and allow users to page from user telephones.

Detailed description of Loudspeaker Paging

This section provides a detailed description of the Loudspeaker Paging feature.

You can administer up to nine separate zones, or sets of loudspeakers on Avaya Communication Manager. Thus you can make an announcement to one group or location without disturbing people who do not need to hear the announcement. Auxiliary trunks connect the speakers in each zone to ports on an auxiliary trunk circuit pack.

Types of Paging

Communication Manager offers two types of loudspeaker paging. You can use each separately, and you can also use both together.

- Voice paging

With voice paging, users can 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, or a chime code to each extension. The chime code assigned to that extension plays over the speakers when that extension is paged.

Chime paging is sometimes called Code Calling Access.

Deluxe Paging

With standard voice paging, users page by dialing the Trunk Access Code assigned to the zone they want to page. If users have an active call, the users must manually put the call on hold or park the call before they dial the TAC.

When you enable deluxe paging, users can automatically park an active call when they use the voice paging feature.

Users with multiappearance telephones

The following description applies only to systems with deluxe paging. To page and park an active call simultaneously, users with a multiappearance 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 presses **Conference** instead of **Transfer**, they are conferenced with the parked caller and both are connected in a three-way conference with the paged user when that user responds. This is called Meet-Me Conferencing.

If the paging user does not 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 parties are connected in a three-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, the user hears a confirmation tone, and the active call is automatically parked on their extension. When the paged party answers the call. The paged party is connected to the paging party. The paging party can then transfer the call to the calling party.

Chime paging

To page a user, dial the TAC for a zone, and then chime the extension of the person that 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

Communication Manager requires a separate port for each paging zone, and supports a maximum of nine zones. If you have more than nine zones or do not 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. In this case, the person paged can speak directly to the pager over the loudspeaker.

Restrictions on loudspeaker paging

These restrictions apply to both voice, deluxe voice, and chime paging:

- A paging call cannot be placed on hold, included in a conference call, or transferred. Also, ringback queuing does not 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 telephone does not hear a call-waiting tone if the user gets a call while paging.

- Listed Directory Number (LDN) and Direct Inward Dialing (DID) 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.

Hardware requirements for Loudspeaker Paging

The Loudspeaker Paging feature requires the following hardware:

- None

Administering Loudspeaker Paging

The following steps are part of the administration process for the Loudspeaker Paging feature:

- [Setting up Voice Paging over loudspeakers](#)
- [Setting up Chime Paging over Loudspeakers](#)
- [Assigning a chime page code to an individual extension](#)

This section describes:

- Any prerequisites for administering the Loudspeaker Paging feature
- The screens that you use to administer the Loudspeaker Paging feature
- Complete administration procedures for the Loudspeaker Paging feature

Prerequisites for administering Loudspeaker Paging

You must complete the following actions before you can administer the Loudspeaker Paging feature:

- The server that runs Communication Manager must have one or more auxiliary trunk circuit packs with enough available ports to support the number of paging zones that you define. Each paging zone requires one port.

See the *Hardware Guide for Avaya™ Communication Manager* for information on specific circuit packs.

- To enable deluxe paging, ensure that the Deluxe Paging and Call Park Timeout to Originator field is set to y in the **Feature-Related System Parameters** screen.

Screens for administering Loudspeaker Paging

Screen name	Purpose	Fields
<i>Loudspeaker Paging</i>	Set voice paging over loudspeakers.	• Voice Paging Timeout • Port • Voice Paging - COR • Voice Paging - TAC • Location
<i>Loudspeaker Paging</i>	Set chime paging over loudspeakers	• Code Calling Playing Cycles • Port • Voice Paging - TAC • Voice Paging - COR • Location • Code Calling - TAC
<i>Code Calling Ids</i>	Set chime paging over loudspeakers	• Ext • Id

Setting up Voice Paging over loudspeakers

To set up voice paging over loudspeakers:

- 1 Type **change paging loudspeaker**. Press **Enter**.

In this example, you set up voice paging for an office with five zones. You allow users to page all five zones at once, and assign a class of restriction (COR) of 1 to all zones.

The system displays the *Loudspeaker Paging* screen ([Figure 196, Loudspeaker Paging screen](#), on page 759).

Figure 196: Loudspeaker Paging screen

LOUDSPEAKER PAGING

CDR? y

Voice Paging Timeout (sec): 30

Code Calling Playing Cycles:

PAGING PORT ASSIGNMENTS									
Zone	Port	Voice Paging			Code Calling			Location:	
		TAC	COR	TN	TAC	COR	TN		
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 the maximum number of seconds that a page can last.
In this example, the paging party is disconnected after 30 seconds.
- 3 In the **Port** field for Zone 1, type the port number that is assigned to the auxiliary trunk circuit pack to this zone.
In this example, the port number is **01C0501**.
- 4 In the **Voice Paging - TAC** field, type the value the for Trunk Access Code (TAC) that users dial to page this zone.
In this example, the value is **301**.
- 5 In the **Voice Paging - COR** field, type the class of restriction (COR) for this zone. You can assign different CORs to different zones.
In this example, the COR is **1**.
- 6 In the **Location** field on the Zone 1 row, type a descriptive name for the zone. Use this name to help you remember the corresponding physical location.
In this example, the zone name is **Reception area**.
Repeat steps 4 through 6 for each zone.
- 7 In the **ALL** row of the **Voice Paging- TAC** field in, type **310** and **1** in the **Voice Paging - COR** field.
When you complete this row, you allow users to page all zones at once. You do not have to assign a port to this row.
- 8 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.

Setting up Chime Paging over Loudspeakers

To set up chime paging for a clothing store with three zones.

- 1 Type **change paging loudspeaker**. Press **Enter**.

You allow users to page all zones at once, and you l assign a COR of 1 to all zones.

The system displays the **Loudspeaker Paging** screen ([Figure 197, Loudspeaker Paging screen](#), on page 760).

Figure 197: Loudspeaker Paging screen

LOUDSPEAKER PAGING									
								CDR? y	
								Voice Paging Timeout (sec):	
								Code Calling Playing Cycles: 2	
PAGING PORT ASSIGNMENTS									
Zone	Port	Voice Paging			Code Calling			Location:	
		TAC	COR	TN	TAC	COR	TN		
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 the number of times that a chime code plays when someone places a page.
In this example, the number of times is **2**.
- 3 In the **Port** field for Zone 1, type the port number of the auxiliary trunk circuit pack that is assigned to this zone.
In this example, the port number is **01A0301**.
- 4 In the **Code Calling - TAC** field, type the Trunk Access Code (TAC) users dial to page this zone. You cannot assign the same trunk access code to more than one zone.
In this example, the trunk access code is **80**.
- 5 In the **Code Calling - COR** field, type the COR number that is assigned to this zone. You can assign different classes of restriction to different zones
In this example, the COR is **1**.
- 6 In the Zone 1 row of the **Location** field, type a descriptive name for the zone. Use this name to help you remember the corresponding physical location.
In this example, the location is **Men's Department**.
Repeat steps 4 through 6 for zones 2 and 3.

- 7 In the **ALL** row of the Code Calling - TAC field, type **89** and **1** in the Code Calling - COR field.

When you complete this row, you allow users to page all zones at once. You do not have to assign a port to this row.

- 8 Press **Enter** to save your changes.

Assigning a chime page code to an individual extension

To assign chime codes to an individual extension:

- 1 Type **change paging code-calling-ids** and press **Enter**.

The system displays the **Code Calling IDs** screen ([Figure 198, Code Calling IDs screen](#), on page 761).

Figure 198: Code Calling IDs screen

ID ASSIGNMENTS		CODE CALLING IDS									
Id	Ext	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 In the **Ext** field, type the first extension, **2130**, and **Id 111**.
In this example, the first extension is **2130**. Each code Id defines a unique series of chimes.
- 3 To assign chime codes to the remaining extensions type an extension number on the line following each code Id.
You can assign a different chime code to as many as 125 extensions.
- 4 Press **Enter** to save your changes.

Reports for Loudspeaker Paging

The following reports provide information about the Loudspeaker Paging feature:

- None

Considerations for Loudspeaker Paging

This section provides information about how the Loudspeaker Paging feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Loudspeaker Paging under all conditions. The following considerations apply to Loudspeaker Paging:

- None

Interactions for Loudspeaker Paging

This section provides information about how the Loudspeaker Paging feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Loudspeaker Paging in any feature configuration.

- **Bridged Call Appearance**
If a parked call includes a shared terminating extension group, a shared Personal Central Office Line (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 usually 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 (PCOL)**
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 (TEG)**
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.

Troubleshooting Loudspeaker Paging

This section lists the known or common problems that users might experience with the Loudspeaker Paging feature.

Problem	Possible cause	Action
Users cannot page.	The attendant has control of the trunk group.	Deactivate attendant control.
Calls to an extension are heard over the loudspeakers.	The extension might have been forwarded to a trunk access code used for paging.	Deactivate call forwarding or change the extension to which calls are forwarded.

Malicious Call Trace

Use the Malicious Call Trace (MCT) feature to track malicious calls. With MCT, you can define a group of telephone users. These users notify other users in the group when they receive a malicious call. These users can then retrieve information that is related to the call. You can identify the source of the malicious call, or provide information to personnel at an adjacent media server or switch to complete the trace. You can use MCT to record the malicious call.

You allow users in the group to activate MCT and control malicious call trace. The controlling telephone user, or controller, receives the information that MCT collects on the call.

MCT does not work with the Avaya S8300 Media Server at this time.

Detailed description of Malicious Call Trace

This section provides a detailed description of the Malicious Call Trace (MCT) feature.

- Activating MCT
- Deactivating MCT
- Using the MCT voice recorder
- Controlling MCT
- Administering MCT for ISDN notification

Activating MCT

To activate MCT while on an active malicious call, perform one of the following actions:

- Push the MCT-Activate feature button.
- Place the call on hold, get a second call appearance, and dial the Feature Access Code (FAC) for an MCT-Activate. When you hear the dial tone, dial your own extension, press the pound key (#), or wait for a 10-second timeout.
- Signal another user in the defined group to activate MCT. This user activates MCT, waits for the dial tone, and dials the extension of the call's recipient.
- Inform a controller. The controller can request another media server to continue to trace the call.

The servers or switches must be tandemed. The controller on the first server supplies the trunk member port ID to be traced. The controller on the second server activates MCT, presses the star key (*), and then dials 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 might be entered as 0130401.

Once a user activities MCT, MCT collects information on the call, and alerts users in the group. The alert is not a call. The alert is not affected by queues at the user's terminal. If an MCT Voice Recorder is connected, MCT starts to record the conversation.

NOTE:

Any Bridging, Conference, or Intrusion tone are temporarily removed while the MCT Voice Recorder connects.

Deactivating MCT

To deactivate MCT, the controller dials the MCT-Deactivate FAC. Deactivation frees any blocked resources involved in the trace. When all parties hang up, system disconnects the MCT Voice Recorder.

Using the MCT voice recorder

The MCT Voice Recorder is any type of audio recorder. The recorder can be a standard audio cassette player that you can control by means of the Avaya Auxiliary Trunk board.

To record the call, manually place the MCT Voice Recorder in Record MCT mode. The telephone user then activates the MCT feature which applies power to the recorder (using the connected Auxiliary Trunk's control signal interface).

Controlling MCT

The first controlling telephone to respond to an MCT alert becomes the controller for the call. Alerting on any other controlling terminals stops.

MCT begins to trace an incoming call when the controller dials the 3-digit equipment location of the incoming trunk.

During alerting, the display of the controller shows the message `Malicious Call Trace Request`. While this message displays, there is no information on incoming calls.

When the controller pushes the MCT-Control button, the system displays information that identifies the called party. When the controller pushes the button again, the system displays the remaining MCT information.

MCT collects and displays three types of calling information:

- If the call originates inside the system or on the same node within a Distributed Communications System (DCS) network, the calling number displays.
- If the call originates outside the system and an Integrated Services Digital Network (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 media server.
- For all calls, the software displays the called number, the activating number, whether the call is active, and identification of any other parties who are on the call.

Administering MCT for ISDN notification

This section describes how to administer the ISDN MCT notification for an ISDN trunk group, including public-ntwrk, tandem, tie, or access trunk groups.

Display the **Optional Features** 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 servers running Communication Manager, set the **Country Protocol** field to **1**, and the **Protocol Version** field to **a**. Avaya recommends these settings if you use DCS features in the private network.
- For a private network of servers running that are running Communication Manager, set the **Peer Protocol** field to **q-sig**.

The **Peer Protocol** field appears on the **DS1** screen when the **Signaling Mode** field is **isdn-pri**, the **Connect** field is **pbx**, and the **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 the **Country Protocol** field to **2**.
- For a private network of servers running Communication Manager, set **Interface** to **peer-master** or **peer-slave**.

Hardware requirements for Malicious Call Trace

The Malicious Call Trace feature requires the following hardware:

- None

Administering Malicious Call Trace

The following steps are part of the administration process for the Malicious Call Trace feature:

- None

Screens for administering Malicious Call Trace

Screen name	Purpose	Fields
<i>Class of Restriction</i>	Set access to MCT.	
<i>Station</i>	Define feature button assignments.	• Mct-act • Mct-contrl

Reports for Malicious Call Trace

The following reports provide information about the Malicious Call Trace feature:

- None

Considerations for Malicious Call Trace

This section provides information about how the Malicious Call Trace feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Malicious Call Trace under all conditions. The following considerations apply to Malicious Call Trace:

- If the originator of the call disconnects, the system discontinues MCT. However, if the recipient of the call disconnects, the system does not discontinue MCT.
- Except for Emergency Access to the Attendant, features that usually display information do not do so on a controlling telephone. Except for the display of information, these features function normally until MCT deactivates.
- Do not use Feature Access Code (FAC) to activate MCT because this requires too much time.
- 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 server failure.
- When you direct a trace to an adjacent server or switch, consider the following possibilities:
 - The malicious caller might hear a warning tone as a result of the intrusion.
 - You can lose continuity on the trace because the person activating MCT on the second server might 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 media server and a cross-reference of system-trunk port numbers. This can also include the DS1 channel number, if appropriate. But the controller might not have the trunk equipment locations at the connecting media server. Ensure that they have this information.
- The following actions 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 using the MCT History report.

Interactions for Malicious Call Trace

This section provides information about how the Malicious Call Trace (MCT) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Malicious Call Trace in any feature configuration.

- **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 using 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 timeout.

- **Conference**

A user can use conferencing to place a malicious caller on hold. The user can start 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 is performed by telephones within the same media server.

- **Distributed Communications System (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 is performed by telephones within the same DCS node.

- **Emergency Access to the Attendant**

Usually, 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. It can also take place on any trunk on the call with an ISDN private network trunk and 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**

Communication Manager attempts to activate Make-Busy or Position-Busy for phones or consoles that activate MCT-Control. If a user has a 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 until 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 starts 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 (TAC)**

To activate MCT for a 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 server or switch to terminate on another server or switch.
- **Trunk Groups**

If a PCOL is involved in an MCT, then Communication Manager might hold up the trunk until the MCT deactivates.

Manual Message Waiting

Use the Manual Message Waiting feature to cause the Manual Message Waiting button lamp at another user telephone to light.

Detailed description of Manual Message Waiting

This section provides a detailed description of the Manual Message Waiting feature.

A user presses the designated button to light the Manual Message Waiting button lamp at the telephone of another user telephone. Both the telephones must be multiappearance telephones. To turn the lamp off, either telephone user can press the Manual Message Waiting button.

You can administer Manual Message Waiting for pairs of telephones only. These telephones might be used by two people who share the same job function and often take calls for one another. The Manual Message Waiting feature is also useful in situations where one person usually answers calls for a second person, such as an administrative assistant might do for a manager. The administrative assistant can press a Manual Message Waiting button to signal the manager that a call must be answered. The manager can answer the call or press a Manual Message Waiting button to indicate that the administrative assistant should handle the call.

Hardware requirements for Manual Message Waiting

The Manual Message Waiting feature requires the following hardware:

- None

Administering Manual Message Waiting

The following steps are part of the administration process for the Manual Message Waiting feature:

- [Assigning the Manual Message Waiting feature button](#)

This section describes:

- Any prerequisites for administering the Message Waiting feature
- The screens that you use to administer the Message Waiting feature
- Complete administration procedures for the Message Waiting feature

Prerequisites for administering Manual Message Waiting

You must complete the following actions before you can administer the Message Waiting feature:

- None

Screens for administering Manual Message Waiting

Screen name	Purpose	Fields
Station (multiappearance)	Assign the Manual Message Waiting feature button for a user.	Button/Feature Button Assignments - man-msg-wt

Assigning the Manual Message Waiting feature button

To assign a Manual Message Waiting feature button for a user:

- 1 Type **change station *n***, where *n* is the extension of the user to whom you want to assign a Manual Message Waiting feature button. Press **Enter**.

The system displays the **Station** screen for the extension that you requested ([Figure 199, Station screen](#), on page 772).

Figure 199: Station screen

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	7:	
2: call-appr	8:	
3: call-appr	9:man-msg-wt	
4:	10:dn-dst	
5:	11:goto-cover	
6:	12:send-calls Ext:	

- 2 Page through the screens until you see the **Button Assignments** area.
- 3 In the **Button Assignments** area, type **man-msg-wt** next to the button that you want the user to use for Manual Message Waiting.
- 4 Press **Enter** to save your changes.

Reports for Manual Message Waiting

The following reports provide information about the Manual Message Waiting feature:

- None

Considerations for Manual Message Waiting

This section provides information about how the Message Waiting feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Manual Message Waiting under all conditions. The following considerations apply to Manual Message Waiting:

- None

Interactions for Manual Message Waiting

This section provides information about how the Message Waiting feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Message Waiting in any feature configuration.

- None

Manual Signaling

Use the Manual Signaling feature to signal another user.

Detailed description of Manual Signaling

This section provides a detailed description of the Manual Signaling feature.

With the manual signaling feature, one user can signal another user. When a user presses the manual signaling button, the other user hears a 2-second ring. The status lamp of the user who presses the button lights for two seconds.

If the telephone of the intended recipient of the signal is already alerting, the system:

- Does not generate the 2-second ring
- Causes the manual signaling button lamp of the user who presses the button to flicker briefly

Hardware requirements for Manual Signaling

The Manual Signaling feature requires the following hardware:

- None

Administering Manual Signaling

This section contains the screens for administering for the Manual Signaling feature.

Screens for administering Manual Signaling

Screen name	Purpose	Fields
<i>Station - multiappearance</i>	Assign a manual signaling button to a user telephone, and specify the extension that rings when the user presses the button.	Button/Feature Button Assignments • signal (Ext:___)

End-user procedures for Manual Signaling

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Manual Signaling:

- Using Manual Signaling

To use manual signaling:

- Press the manual signaling button.

Reports for Manual Signaling

The following reports provide information about the Manual Signaling feature:

- None

Considerations for Manual Signaling

This section provides information about how the Manual Signaling feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Manual Signaling under all conditions. The following considerations apply to Manual Signaling:

- None

Interactions for Manual Signaling

This section provides information about how the Manual Signaling feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Manual Signaling in any feature configuration.

- Data Modules

If you administer a manual signaling button for a data module, the system denies any attempt to activate the button.

- Vector Directory Number (VDN)

A manual signaling button cannot point to a VDN.

Meet-me Conference

Use the Meet-me Conference feature to set up a dial-in conference of up to six parties. The Meet-me Conference feature uses Call Vectoring to process the setup of the conference call.

Detailed Description of Meet-me Conference

This section provides a detailed description of the Meet-me Conference feature.

With the Meet-me Conference feature, a station user can host a dial-in conference of up to six parties. You can set up the Meet-me Conference extension to require an access code. If you administer an access code, all parties must correctly enter the access code to join the conference. If the extension is one of your Direct Inward Dialing (DID) numbers, any internal or remote access users, or external parties, can dial the Meet-me Conference extension.

When a caller dials into a Meet-me Conference, the system plays an announcement. The announcement tells the caller to enter an access code, if an access code is required. If the caller enters the correct access code, the caller is added to the conference. If no other parties are on the conference, an announcement tells the caller that he is the first caller to join the conference. If other parties are already on the call, an announcement tells the caller that he is joining a conference that is already in progress. All parties who are already on the conference and the newly added party hear an entry tone. When a party drops out of the conference, the remaining parties hear an exit tone.

Hardware requirements for Meet-me Conference

The Meet-me Conference feature requires the following hardware:

- **Announcement Board**
Integrated announcements reside on a circuit pack in the switch carrier. You can store multiple announcements on the circuit pack, up to the system capacity.

Administering Meet-me Conference

The following steps are part of the administration process for the Meet-me Conference feature:

- [Creating or changing a Meet-me Conference vector](#)
- [Creating a Meet-me Conference VDN](#)

This section describes:

- Any prerequisites for administering the Meet-me Conference feature
- The screens that are required to administer the Meet-me Conference feature
- Complete administration procedures for the Meet-me Conference feature

Prerequisites for administering Meet-me Conference

You must complete the following actions before you can administer the Meet-me Conference feature:

- On the **Optional Features** screen, ensure that the G3 Version field is set to **V11** or later. If the G3 Version field is not set to **V11** or later, your system is not enabled for the Meet-me Conference feature. Contact your Avaya representative before you continue with this procedure.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

- On the **Optional Features** screen, click **Next** until you see the Enhanced Conferencing field. Ensure that this field is set to **y**. If the Enhanced Conferencing field is set to **n**, your system is not enabled for the Meet-me Conference feature. Contact your Avaya representative before you continue with this procedure.

Screens for administering Meet-me Conference

Screen Name	Purpose	Fields
Optional Features	Ensure that you have Communication Manager version 1.3 (V11) or greater.	G3 Version
	Ensure that Enhanced Conferencing is enabled.	Enhanced Conferencing
Call Vector	Create a Meet-me Conference vector.	All
Vector Directory Number	Set up a VDN for Meet-me Conference.	- Extension - Name - Vector Number - Meet-me Conferencing - Conference Access Code - Conference Controller
Meet-me Vector Directory Number	View a list of existing Meet-me Conference VDNs.	All

Creating or changing a Meet-me Conference vector

To create a vector for a Meet-me Conference:

- Type **change vector x**, where **x** is the number of the vector that you want to create or change. Vector numbers must be between 1 and 256. Press **Enter**.

The system displays the **Call Vector** screen ([Figure 200](#)).

Figure 200: Call Vector screen

change vector 129		CALL VECTOR		Page 1 of 3	
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 _____					

- 2 In the **Meet-me Conf** field, type **y** to designate the vector as a Meet-me Conference vector.
- 3 Use the numbered fields to create or change a Meet-me Conference vector, as shown in the examples in [Figure 200, Call Vector screen](#), on page 779 and [Figure 201, Call Vector screen](#), on page 779. See [Options for creating vector steps](#) and [How the vector processes a call](#) for more information on how to create a vector.

Figure 201: Call Vector screen

change vector 90		CALL VECTOR		Page 1 of 3		SPE A	
Number: 90		Name: Enh Conf Vec					
		Attendant Vectoring? n	Meet-me Conf? y	Lock? y			
Basic? y	EAS? n G3V4 Enhanced? n	ANI/II-Digits? n	ASAI Routing? n				
Prompting? y	LAI? n G3V4 Adv Routs? n	CINFO? n BSR? n	Holidays? n				
01 collect	6 digits after announcement 12340						
02 goto	step 6 if digits = meet-me-access						
03 collect	6 digits after announcement 12341						
04 goto	step 6 if digits = meet-me-access						
05 disconnect	after announcement 12342						
06 goto	step 11 if meet-me-idle						
07 goto	step 14 if meet-me-full						
08 announcement	12343						
09 route-to	meetme						
10 stop							
11 announcement	12344						

Figure 202: Call Vector screen

change vector 90		CALL VECTOR		Page 2 of 3 SPE A	
Number: 90		Name: Enh Conf Vec			
		Attendant Vectoring? n		Meet-me Conf? y	
Basic? y		EAS? n G3V4 Enhanced? n		ANI/II-Digits? n ASAI Routing? n	
Prompting? y		LAI? n G3V4 Adv Routs? n		CINFO? n BSR? n Holidays? n	
12 route-to		meetme			
13 stop					
14 disconnect		after announcement 12345			
15 stop					
16					
17					
18					
19					
20					
21					
22					

4 Press **Enter** to submit the vector.

NOTE:

If a vector allows a new party to join a conference immediately, and the party is an H.323 IP trunk user, the caller might not have a talk path with the other parties in the conference. To prevent this situation, include in the vector a short delay before a new party joins the Meet-me Conference. This delay can be a step to collect digits, a 1-second delay, or an announcement. Since Meet-me vectors are always configured with announcements and digit collections, this situation is rarely an issue.

Options for creating vector steps

- **collect**

When the **Meet-me Conf** field is enabled, the **collect** vector step collects the next six digits. The vector step then uses those digits as the access code for a Meet-me Conference call. See vector steps 1 and 3 in the example in [Figure 201, Call Vector screen](#), on page 779.

- **goto**

The **goto** vector step has three conditions:

- *meet-me-idle*

The meet-me-idle condition routes the first caller who accesses a Meet-me Conference to the conference call. An announcement step that tells the caller that he is the first party to join the conference can be played. See vector steps 6 and 11 in the example in [Figure 201, Call Vector screen](#), on page 779.

- *meet-me-full*

- The meet-me-full condition is used when the Meet-me Conference already has the maximum of six parties on the call. See vector steps 7 and 14 in the example in [Figure 202, Call Vector screen](#), on page 780.

- *meet-me access*

- The meet-me access condition ensures that the access code is valid. If the access code that the caller dials is the same as the access code that is administered for the VDN, vector processing continues. See vector steps 2 and 4 in the example in [Figure 201, Call Vector screen](#), on page 779.

- **route-to**

The **route-to** vector step has one condition:

- *meetme*

This condition adds the caller to the Meet-me Conference call, and all parties on the call hear an entry tone. The *meetme* condition is valid when the caller enters the correct access code, and the number of parties who are on the call already is less than six. See vector steps 9 and 12 in the example in [Figure 202, Call Vector screen](#), on page 780.

If the **route-to meetme** step fails, vector processing stops, and the caller hears a busy tone.

How the vector processes a call

This section describes what occurs when a caller dials the Meet-me Conference telephone number that is managed by the vector in [Figure 201, Call Vector screen](#), on page 779 and [Figure 202, Call Vector screen](#), on page 780.

The caller hears announcement 12340. Announcement 12340 says, “Welcome to the Meet-me Conferencing service. Enter your conference access code.” The caller enters the access code 937821. The **collect** vector step 1 collects the access code digits. If the access code is valid, the vector processing continues with vector step 6.

If the access code is invalid, vector step 3 plays announcement 12341. Announcement 12341 says, “The access code you entered is invalid. Please enter the access code again.”

If the caller enters the wrong access code again, vector step 5 plays announcement 12342. Announcement 12342 says, “This access code is invalid. Please contact the conference call coordinator to make sure you have the correct conference telephone number and access code. Goodbye.” The caller is disconnected.

Vector step 6 is only valid for the first caller into the Meet-me Conference. The *meet-me-idle* condition routes the first caller to announcement 12344, according to vector step 11. The announcement says, “You are the first party to join the call.” The system then routes the caller to the Meet-me Conference call by vector step 12, and vector processing stops.

Vector step 7 is used when the Meet-me Conference maximum of six parties are already on the call. The *meet-me-full* condition disconnects the caller after announcement 12345 plays, according to vector step 14. Announcement 12345 says, “This Meet-me Conference is filled to capacity. Please contact the conference call coordinator for assistance. Goodbye.”

If a caller enters the correct access code, is not the first caller, and the conference is not full, processing continues with vector step 8. Announcement 12343 plays. The announcement says, “Your conference call is already in progress.” The system then routes the caller to the Meet-me Conference call by vector step 9, and vector processing stops.

When a caller enters the conference, all parties on the call hear an entry tone. When a party drops out of the conference, the remaining parties hear an exit tone.

Creating a Meet-me Conference VDN

To create a Meet-me Conference vector directory number (VDN):

- 1 Type **add vdn next**, or **add vdn *n***, where *n* is the extension that you want to use for the VDN. Note that if the VDN extension is one of your Direct Inward Dialing (DID) numbers, external users can access the conference VDN. If the VDN extension is not part of the DID block, only internal callers on the network, or remote access callers, can access the conference. Press **Enter**.

The system displays the **Vector Directory Number** screen ([Figure 203, Vector Directory Number screen](#), on page 782).

Figure 203: Vector Directory Number screen

VECTOR DIRECTORY NUMBER
Extension: 36090
Name: Meet-me Conference VDN
Vector Number: 90
Meet-me Conferencing? y
COR: 1
TN: 1

- 2 In the **Extension** field, type the extension for the VDN. Or, if you typed the extension in Step 1 as part of the **add vdn** command, the system automatically displays the extension in the **Extension** field.
- 3 (Optional) In the **Name** field, type a name of up to 27 characters to identify this VDN.
- 4 In the **Vector Number** field, type the number for this vector. Or, if you typed **add vdn next** in Step 1, the system automatically displays the next available vector number in the **Vector Number** field.
- 5 In the **Meet-me Conferencing** field, type **y**.
- 6 Click **Next** to go to the **Meet-me Conference Parameters** page of the **Vector Directory Number** screen.

The system displays the **Meet-me Conference Parameters** screen ([Figure 204, Meet-me Conference Parameters screen](#), on page 782).

Figure 204: Meet-me Conference Parameters screen

VECTOR DIRECTORY NUMBER
MEET-ME CONFERENCE PARAMETERS:
Conference Access Code: 937821
Conference Controller: 80214

- 7 In the **Conference Access Code** field, assign a six-digit access code for the Meet-me Conference. Avaya recommends that you always assign an access code for a Meet-me Conference. However, if you do not want to assign an access code, leave the **Conference Access Code** field blank. Once you assign an access code, the system displays an asterisk (*) in this field for subsequent change, display, or remove operations by all users except the *init* superuser login. An administrator who uses the *init* login sees the actual access code instead of an asterisk.
- 8 In the **Conference Controller** field, type the extension of the person who is responsible to control or change the Meet-me Conference access code. If you type an extension, a user at that extension can use a feature access code (FAC) to change the access code. If you leave the **Conference Controller** field blank, any station user to whom console permissions are assigned can change the access code. Remote access users can also use a FAC to change a Meet-me Conference access code.
- 9 Press **Enter** to submit the VDN.

For more information on feature access codes, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

To view a list of existing Meet-me Conference VDNs:

- Type **list meet-me-vdn**. Press **Enter**.

The system displays the **Meet-me Vector Directory Number** screen ([Figure 205, Meet-me Vector Directory Number screen](#), on page 783).

An asterisk (*) in the Access Code field indicates that an access code is assigned for that Meet-me VDN. If the Access Code field is blank, no access code is assigned. An administrator who uses the *init* login sees the actual access code instead of an asterisk.

Figure 205: Meet-me Vector Directory Number screen

list meet-me-vdn						
MEET-ME VECTOR DIRECTORY NUMBER						
Name	Ext	Access Code	COR	TN	Vec Num	Control Ext
Secure Meet-me Conference	4000	*	1	1	1	
Nonsecure Meet-me Conference	4006		1	1	2	84590

End-user procedures for Meet-me Conference

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures:

- [Accessing a Meet-me Conference as an attendee](#)
- [Changing a Meet-me Conference access code](#)
- [Using Selective Conference Party Display, Drop, and Mute](#)

Accessing a Meet-me Conference as an attendee

To access a Meet-me Conference as an attendee:

- 1** Dial the Meet-me Conference telephone number. If an access code is required, a recorded announcement tells you to enter the access code.
- 2** Enter the Meet-me Conference access code.

The system verifies the access code, and connects you to the Meet-me Conference. You hear an entry tone when you join the conference.

If the maximum of six parties are already connected to the Meet-me Conference, a recorded announcement tells you that the Meet-me Conference is full.

Changing a Meet-me Conference access code

Both local and remote access telephone users can change access codes. However, an access code must first be administered through system administration before a telephone user can change a code. Access codes must be removed through system administration.

To change a Meet-me Conference access code:

- 1** Dial the feature access code (FAC) for Meet-me Conference.
- 2** When you hear dial tone, dial the Meet-me Conference VDN, and then press the pound key (#).
- 3** Dial the current access code, and then press the pound key (#).
- 4** When you hear dial tone, dial the new access code, and then press the pound key (#).
- 5** Dial the new access code again, and then press the pound key (#).
- 6** When you hear the confirmation tone, hang up the telephone.

If any errors occur during this operation, you hear intercept tone, and the access code is not changed. You must start over.

Using Selective Conference Party Display, Drop, and Mute

You can use this feature from a digital display station or from an attendant console. In this example, stations A, C, and D are on a conference call. Caller B is on the conference call using an outside trunk or a cellular telephone.

- 1** Station A presses the Conference Display button.
The LED for the Conference Display button lights. The station displays the name and the number for station C, if this information is available.
- 2** Station A can press the Conference Display button repeatedly to cycle through all parties on the call.
- 3** When the name and the number for station C displays on Station A, Station A presses the Drop button, or the Forced Release button on the attendant console.
Station C is dropped from the conference call. Callers A, B, and D remain on the conference call. The display for station A now shows one of the other parties on the call.
- 4** Caller B from an outside trunk puts the conference call on hold. This action adds music-on-hold to the conference call.
- 5** Station A presses the Conference Display button until the station displays the name and the number for Caller B.
- 6** Station A presses the Far End Mute button.

NOTE:

Far-end Mute can only be activated for trunks. It cannot be activated for stations.

Caller B is put into “listen-only” mode, and the music-on-hold is removed from the conference call. The Station A display indicates that the outside trunk call (B) is muted.

- 7** Caller B selects the conference call appearance to return to the conference call. To exit “listen-only” mode, the outside trunk caller presses the pound key (#).
Caller B is again active on the conference call.
- 8** Station A presses the Exit button, or the Normal button on the attendant console.
Station A returns to normal mode. Conference 3 displays on Station A. If station A is inactive for 60 seconds, Station A returns automatically to normal mode.

If the Selective Conference Party Mute feature is activated without the knowledge of the muted party, that party might think that a problem exists with the connection when no one responds on the conference call. Users must be instructed about this new feature, and how to return to the call if the users are muted. If the muted party does not know how to return to the call, another user on the conference call can use the Far End Mute button to unmute the party.

Rotary telephone users who are muted by way of the Selective Conference Party Mute feature cannot add themselves back into the conference call. Another user on the conference call must use the Far End Mute button to unmute the party.

Reports for Meet-me Conference

The following reports provide information about the Meet-me Conference feature:

- None.

Considerations for Meet-me Conference

This section provides information about how the Meet-me Conference feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Meet-me Conference feature under all conditions.

- Attendant Intrusion

An attendant can intrude on a station that is part of a Meet-me Conference, as long as the maximum number of conference parties has not been reached.

- Bridged Appearances

Bridged appearance users can be added to a Meet-me Conference, and count against the total number of conference parties. As bridged appearance users join a Meet-me Conference, the users are not prompted to provide an access code. Bridged appearance users also receive no announcements, nor are entry or exit tones applied when the users are added to or dropped from the call. Avaya assumes that this scenario would only take place with the knowledge of the user of the appearance that originally dialed into the Meet-me Conference. That user could use the Exclusion feature to prohibit the bridged user from being a part of the Meet-me Conference call.

- Busy Verification

If the system maximum of stations in a conference call is five or less, a station or an attendant can verify another station that is part of a Meet-me Conference. If the system maximum of stations in a conference call is six, a station or an attendant can verify another station that is part of a Meet-me Conference, as long as the maximum number of conference parties is not reached.

- Call Vectoring

If a Meet-me Conference VDN is administered to use a vector that has no steps, the call attempt is dropped, and a vector event is generated.

- Capacity issues

A Meet-me Conference call can have a maximum of six parties on the call. Additional parties cannot join a Meet-me Conference once the maximum of six parties is reached.

- Changing vector types

To change a Meet-me Conference vector to a non Meet-me Conference vector, you must first remove all vector steps. To change a non Meet-me Conference vector to a Meet-me Conference vector, you must first remove all vector steps.

- Class of Service (COS)

A station user must have console permissions Class of Service to change a Meet-me Conference access code from his own telephone.

- **Conference**

Parties in a Meet-me Conference call can use the Conference feature to add other parties up to the system conference limit. When the parties are added to the call, the “entry” tone is not given. Two or more Meet-me Conference calls cannot be conferenced together.
- **Conference Tone**

The purpose of the conference tone feature is to ensure that no one can be added to a call without the knowledge of the other parties. The Meet-me Conference already has entry and exit tones for all parties that enter the conference through vector processing. If one of the parties conferences in another user through the station Conference feature, the other parties are not aware of the additional party because no entry tone is played. In this scenario, the conference tone should be played if required as part of the system parameters administration.
- **Conference/Transfer Toggle/Swap feature limitations**

The Conference/Transfer Toggle/Swap feature is unavailable on analog stations and the attendant console. The attendant console can use the Split Swap feature to perform a similar operation.

The station user who presses the Toggle Swap button must be in the talk state with one of the parties. If the button is pressed during ringback, the system ignores the button push.

Station users must be careful when using the Selective Conference Party Display feature to scroll through the displays. The station hyperactivity feature takes the station out of service if the user repeatedly scrolls through the displays at high enough rates. This action causes the station to be reset, and the user is dropped from the call.
- **Disabling Enhanced Conferencing**

To disable the Enhance Conferencing option on the **System Parameters-Customer Options** screen, you must first remove all Meet-me Conference VDNs and vectors. If you do not remove these VDNs and vectors, the system displays a message that tells you that you must first remove all Meet-me Conference VDNs and vectors.
- **Drop**

No controlling party exists in a Meet-me Conference. Therefore, if a caller on the conference call presses the Drop button, the system ignores the button push, and the last party who joined the conference call is not dropped from the call.
- **Far-end Mute**

Only one trunk on a Meet-me Conference bridge can be far-end muted at any given time.
- **Removing stations**

You cannot remove a station that is administered as a controlling station for a Meet-me Conference VDN unless you first remove the assignment on the VDN. If you attempt to remove a controlling station for a Meet-me Conference VDN, the system displays a message that tells you that you must first remove the extension as conference controller on the VDN form.
- **Security issues**

The Meet-me Conference feature can present a potential security problem. If you assign Meet-me Conference VDNs without access codes, a hacker can tie up Meet-me Conference facilities and keep others from conducting legitimate business. A hacker can also potentially access the system, and use the system to make unauthorized calls. Avaya recommends that you administer access codes, and change the codes regularly to reduce the risk of unauthorized access to the system. If a user tries to change the access code of a Meet-me Conference and is unsuccessful, or uses an invalid access code, the system records an event to the Event Log.
- **Service Observing**

Service Observing by way of the VDN is not allowed for Meet-me Conference VDNs.

- **Transfer**

When transferring a call to a Meet-me Conference, the transfer can be completed during vector processing only when a single party is on soft hold waiting to be transferred. If two or more parties are on soft hold waiting to be transferred, the transfer can only be completed after the party who is initiating the transfer is connected to the Meet-me Conference.

- **Vectoring options**

Attendant Vectoring and Meet-me Conference cannot both be enabled at the same time. If Enhanced Conferencing is enabled, but no other vectoring customer options are enabled, only Meet-me Conference vectors can be assigned. A non Meet-me Conference vector cannot be assigned to a Meet-me Conference VDN. A Meet-me Conference vector cannot be assigned to a non Meet-me Conference VDN. No restrictions exist with regard to vector “chaining” between Meet-me Conference and non Meet-me Conference vectors. When a call interflows from one type of vector processing to another, the call is removed from any queue, if applicable. The call is treated as a new call to vectoring, rather than as a continuation of vectoring.

Interactions for Meet-me Conference

This section provides information about how the Meet-me Conference feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of the Meet-me Conference feature in any feature configuration.

- **Bridged Appearances**

If a station user who is active on a bridged appearance makes a conference or transfer, the user receives the enhanced displays based on the COR of station of the user, and not the station with the primary extension.

- **Call Detail Recording (CDR)**

As parties join a Meet-me Conference, a call record is created, if a record is required by system administration. If a record is required, the called party is the Meet-me Conference VDN number. The duration is the length of time that the party was included in the call. An individual record for each party is generated when the party drops from the call.

One option for recording all calls to Meet-me Conference VDNs is to activate the Intraswitch CDR feature, and populate all system Meet-me Conference VDN numbers. If you use the Intraswitch CDR feature with the Meet-me Conference VDNs, set the condition code to **C** for all call records. If the Intraswitch CDR feature is not active for Meet-me Conference VDNs, the creation and the content of call records depends on the trunk group translations for external callers to the Meet-me Conference. Internal callers to the Meet-me Conference do not generate any records if the Intraswitch CDR feature is not active for either the Meet-me Conference VDN or the calling extension.

- **Consult**

If the covering party is talking to the principal after the covering party presses the Consult button, the covering party can use the Toggle Swap button to toggle back and forth between the caller and the principal.

- Direct Inward Dialing (DID)

If a Meet-me Conference VDN is one of your DID numbers, external users can access the conference VDN. If the VDN extension is not part of the DID block, only internal callers on the network or remote access callers can access the conference VDN.

- Pull Transfer

If Station A is talking to Station B and Station B begins a transfer to Station C, only Station C can use the Pull Transfer feature to grab the call. Furthermore, if Station B uses the Toggle Swap feature to return to Station A and Station C is now in a soft hold state, Station C cannot use the Pull Transfer feature until Station B toggles back and has Station C in the talk state.

Troubleshooting Meet-me Conference

This section lists the known or common problems that users might experience with the Meet-me Conference feature.

Problem	Possible cause	Action
The conference call drops abruptly for no apparent reason.	The Vector Disconnect Timer field on the System Parameters - Features screen is set to a value that is shorter than the duration of the Meet-me Conference session.	Increase the value in the Vector Disconnect Timer field.
The sound volume is too low.	The affected conference participants connect through international trunks in which central office (CO) loss plans are set for too much loss.	On the Location Parameters screen, adjust the values in the End-to-End total loss (db) in a n-party conference field.

Modem Pooling

Use the Modem Pooling feature to enable switched connections between digital-data endpoints (data modules) and analog-data endpoints by way of pools of acoustic-coupled modems. The analog-data endpoint is either a trunk or a line circuit.

Detailed description of Modem Pooling

This section provides a detailed description of the Modem Pooling feature.

Data transmission between a digital-data endpoint and an analog-data endpoint requires conversion through a modem. Conversion is required because the DCP format that the data module uses is not compatible with the modulated signals of an analog modem. A modem translates DCP format into modulated signals and vice versa.

The Modem Pooling feature provides for the use of pools of integrated-conversion modems and combined-conversion modems.

Integrated-conversion modem pools have functionality that is 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 that are cabled to any TDM-compatible modem. Combined-conversion modem pools can be used with all systems.

When the system needs a modem, the system queries the digital-data module that is associated with the call to determine if the options are compatible with the options that are supported by the modem pools. If the options are incompatible, the originating user receives intercept treatment. If the options are compatible, the system obtains a modem from the appropriate pool. If a modem is unavailable, the user receives reorder treatment.

The system can detect the need 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 the users dial the digital-data endpoint.

The system provides a Hold Time parameter. This parameter specifies the maximum time that 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 that are usually used in 3270 and 2780/3780 applications. Both require 2400 or 4800 bps, half-duplex, synchronous transmission.
- Interactive IBM-TSO applications that use 1200 bps, half-duplex, asynchronous transmissions
- DATAPHONE II switched-network modems that support asynchronous and synchronous communications, and autobaud at 300, 1200, or 2400 bps
- Avaya Communication Manager operating at up to 19.2 kbps
- Different pools with different data-transmission characteristics

Hardware requirements for Modem Pooling

The Modem Pooling feature requires the following hardware:

- None

Administering Modem Pooling

The following steps are part of the administration process for the Modem Pooling feature:

This section describes:

- The screens that you use to administer the Modem Pooling feature

Screens for administering Modem Pooling

Screen name	Purpose	Fields
<i>Modem Pool Group</i>	Define modem pool groups.	All
<i>Feature Access Code</i>	Set data origination access.	Data Origination Access Code
<i>Data Module</i>	Define specific data modules.	All

Reports for Modem Pooling

The following reports provide information about the Modem Pooling feature:

- None

Considerations for Modem Pooling

This section provides information about how the Modem Pooling feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Modem Pooling under all conditions. The following considerations apply to Modem Pooling:

- On data calls between a data module and an analog-data endpoint, Return-to-Voice releases the modem and returns the modem to the pool. The phone user connects to the analog-data endpoint.
- For traffic purposes, the system accumulates data on modem-pooling calls separately from voice calls. Measurements on the pools also accumulate.
- When a telephone 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 an acoustically-coupled modem, the user dials the data-origination access code before the user dials 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 the use of modems from different vendors within a combined pool. Modems from different vendors might have different transmission characteristics.
- When you administer data-transmission characteristics (such as speed, duplex, and synchronization mode), they must be identical to the TDM and the optional modem selections made by the customer.
- Each data call that uses Modem Pooling uses four time slots, instead of two. As a result, heavy usage of Modem Pooling can affect TDM bus-blocking characteristics.
- Tandem servers or switches do not insert a pooled modem. The originating server or switch inserts a pooled modem.

Interactions for Modem Pooling

This section provides information about how the Modem Pooling feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Modem Pooling in any feature configuration.

- **Call Detail Recording (CDR)**
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.

Modem Pooling

Interactions for Modem Pooling

- Digital-multiplexed Interface (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 are used for Modem Pooling to AVD DS1 tie trunks by Data Terminal Dialing or by dialing the feature access code (FAC) for data origination.

Multi-Location Dial Plan

Use the Multi-Location Dial Plan feature to preserve dial plan uniqueness for extensions and attendants when you migrate to a single S8700 distributed network. Without this feature, dial plan uniqueness is not preserved when you migrate from multiple QSIG or distributed communication system (DCS) networks.

Migrating to a single distributed network reduces the number of systems a customer must maintain. With a single network, you administer one system and one dial plan. However, with a single distributed network, some features no longer work transparently across multiple locations as before the migration.

For example, in multiple QSIG or DCS networks, each location of a department store might have its own system. Therefore, the same telephone extension might represent a unique department in all stores. Extension 4321 might be the luggage department in all stores. Store employees at any location can dial 4321 and get the luggage department in the store.

If this customer migrates to a single distributed network, store employees can no longer dial 4321 to reach the local luggage department. The system cannot correctly analyze the digits and route the call to the correct store. Store employees now have to dial a complete telephone extension to reach the luggage department in the store.

This problem is solved with the Multi-Location Dial Plan feature. With the Multi-Location Dial Plan feature, Communication Manager adds the location prefix digits to the front of the dialed number. The software then analyzes the entire dialed string and routes the call based on the administration on the **Dial Plan Parameters** screen.

With the Multi-Location Dial Plan feature, a user can dial a shortened version of an extension instead of having to dial a complete extension. For example, the store employee can continue to dial 4321 instead of having to dial 765-4321 to reach the luggage department.

Without the Multi-Location Dial Plan feature in a single distributed network, a call that is routed to an attendant might not terminate at the local attendant. For example, if a school district migrates to a single distributed network, dialing the attendant access code may not route the call to the local school attendant. With the Multi-Location Dial Plan feature, the system routes the call to the correct attendant.

The Multi-Location Dial Plan feature provides dial plan capabilities that are similar to those of QSIG networks or DCS networks. These capabilities include:

- Extension uniqueness
- Announcements for each location
- Local attendant access
- Local Automatic Route Selection (ARS) code administration

On Linux platforms only, the Multi-Location Dial Plan feature also enhances the dial plan and the **Uniform Dial Plan Table** screen so that users can:

- Dial a shortened version of a telephone extension, and reach the same destination as before the migration
- Dial and reach a centralized local answering point
- Dial a local attendant feature access code (FAC) or ARS FAC, and access the same feature as before the migration

On Linux platforms only, you can also play administered announcements in a language that is based on location.

Detailed description of Multi-Location Dial Plan

This section provides a detailed description of the Multi-Location Dial Plan feature.

With the Multi-Location Dial Plan feature, users in a single distributed network can dial a local extension to reach a number in their area. Before this can happen, the administrator must first retranslate their dial plan and extensions. This feature inserts leading digits from the calling number to the called number for intralocation dialing.

This feature also provides a local centralized answering point for attendants. Finally, with this feature, you can administer multiple feature access codes for ARS and for attendants.

Location prefix

A new value is available in the **Insert Digits** field on the ***Uniform Dial Plan Table*** screen. This new value takes the location prefix of the caller from the ***Locations*** screen. The software adds the location prefix to the front of the called number. The software then analyzes the entire dialed string, and routes the call based on the administration on the ***Dial Plan Parameters*** screen.

- Non-IP telephones and trunks inherit the location number of the cabinet, the remote office, or the media gateway to which they are connected.
- IP telephones and trunks obtain their location number indirectly.
 - On the ***Network Region*** screen, you administer a location number that applies to all telephones in that IP region. If a location field is left blank on the ***Network Region*** screen, an IP telephone obtains its location from the location of the cabinet that contains the CLAN board.
 - IP trunks obtain the location from the location of their associated signaling group.

Either direct administration, which is only possible for remote office signaling groups, or the ways described for IP telephones, determines the location.

If the location prefix is not administered correctly, the software does not route calls correctly. A location prefix is not administered correctly if:

- You administer the prefix with incorrect digits
- The entry on the ***Uniform Dial Plan Table*** screen does not match the length of the location prefix

Feature Access Codes

- The software uses the Auto Route Selection (ARS) - Access Code fields on the **Feature Access Code (FAC)** screen when no value exists in the ARS FAC field on the **Locations** screen. If a value exists in the ARS FAC field on the **Locations** screen, the software uses that location ARS FAC. The Auto Route Selection (ARS) - Access Code fields on the **Feature Access Code (FAC)** screen are ignored.

If you use the ARS FAC field on the **Locations** screen, you lose the ability to administer two ARS codes on the **Feature Access Code (FAC)** screen.

The location ARS FAC is accessible only for calling numbers at locations that are administered with that ARS FAC. The software denies any attempt to use an ARS FAC at a location for which the FAC is not valid.

- The Attendant Access Code field on the **Feature Access Code (FAC)** screen has the same characteristics as the atttd call type on the **Dial Plan Analysis Table** screen.

You can administer only one attendant code. You can administer the attendant code either on the **Dial Plan Analysis Table** screen, or on the **Feature Access Code (FAC)** screen. You cannot administer the attendant code on both screens.

If you want to:

- Administer an attendant access code for different locations, use the Attendant Access Code field on the **Feature Access Code (FAC)** screen.
- Use one attendant code for all locations, administer the code as **atttd** in the Call Type field on the **Dial Plan Analysis Table** screen.

You cannot reuse the value that you use for the atttd call type for any FACs. Nor can you reuse the value for the ARS FAC or Atttd FAC on the **Locations** screen.

Atttd FAC, the attendant code on the **Locations** screen, is accessible only for calling numbers at locations that are administered with that attendant access code. The software denies any attempt to use an attendant code at a location for which the code is not valid.

- Only one attendant or ARS access code is valid for a location. The global ARS access code and the attendant access code, and the dialed string for the attendant call type, are valid for a location only if:
 - A local ARS access code does not exist, or
 - An attendant access code does not exist

Announcements

With multiple QSIG or DCS networks, multiple switches can handle announcements in multiple languages. Without the Multi-Location Dial Plan feature in a single distributed network, the software cannot play announcements in multiple languages.

With the Multi-Location Dial Plan feature, announcement extensions are stored as digit strings. The software routes these digit strings through Uniform Dial Plan (UDP) processing. This processing is similar to dialing a telephone extension. When the software dials a shortened extension for an announcement, the software prepends the location prefix onto the extension. With both the location prefix and the extension, the software can play the announcement to the caller in the proper language.

Invalid announcements

The software provides intercept treatment if the following announcements are invalid:

- DID/Tie/Intercept
- Controlled Outward Restriction Intercept Treatment
- Controlled Termination Restriction (Do Not Disturb)
- Controlled Station to Station Restriction
- Controlled Toll Restriction Intercept Treatment
- Invalid Number Dialed Intercept Treatment

The software skips and does not play the following announcements if the announcements are invalid:

- Analog Busy Auto Callback
- Direct Agent Announcement Extension
- Hospitality

The treatment for an announcement that is invalid during vector processing is the same as if the announcement did not exist. Call processing continues and does not wait. Any previous feedback that precedes the announcement continues.

- An announcement step continues at the next step.
- A wait step that references an announcement continues after any specified wait treatment expires.
- A disconnect step disconnects immediately.
- A collect step continues at the next step.

If the VDN of Origin Announcement (VOA) does not exist, vector processing bypasses the VOA, and delivers the call to the agent.

Maintenance

When you migrate to a single distributed network, you must re-administer your dial plan if you want to maintain extension uniqueness. You can administer the UDP so that users can dial a local extension at an individual location the same as before the migration.

If you administer local ARS FACs on the **Locations** screen, you lose the ability to administer two ARS codes. The ability to administer two ARS codes is currently provided only on the **Feature Access Code (FAC)** screen.

Hardware requirements for Multi-Location Dial Plan

The Multi-Location Dial Plan feature requires the following hardware:

- None

Administering Multi-Location Dial Plan

The following steps are part of the administration process for the Multi-Location Dial Plan feature:

- [Changing extensions](#)
- [Prepending numbers to the dialed string](#)
- [Setting up announcements](#)
- [Setting up a local centralized answering point](#)
- [Setting up multiple feature access codes for attendants](#)
- [Setting up multiple feature access codes for ARS](#)

This section describes:

- Any prerequisites for administering the Multi-Location Dial Plan feature
- The screens that you use to administer the Multi-Location Dial Plan feature
- Complete administration procedures for the Multi-Location Dial Plan feature

Prerequisites for administering Multi-Location Dial Plan

You must complete the following actions before you can administer the Multi-Location Dial Plan feature:

- On the **Optional Features** screen, ensure that the **Multiple Locations** field is set to **y**. If this field is set to **n**, your system is not enabled for the Multi-Location Dial Plan feature. Contact your Avaya representative for assistance.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

- On the **Daylight Savings Rules** screen, ensure that rules for daylight savings time are administered.

To view the **Daylight Savings Rules** screen, type **change daylight-savings-rules**. Press **Enter**.

For a complete description of the many **Optional Features** screens, and the **Daylight Savings Rules** screen, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Multi-Location Dial Plan

Screen name	Purpose	Fields
Optional Features	Ensure that the multiple locations feature is enabled.	Multiple Locations
Daylight Savings Rules	Ensure that rules for daylight savings time are administered.	All

Screen name	Purpose	Fields
<i>Locations</i>	- Set up a FAC to access ARS for a location. - Set up a FAC to access an attendant for a location. - Include the prefix for the specific location.	- ARS FAC - Attd FAC - Prefix
<i>Change Station Extension</i>	Change multiple extensions in the software from one extension to another simultaneously.	All
<i>Uniform Dial Plan Table</i>	Set up your dial plan.	All

Changing extensions

Use the **change extension-station** command to change multiple extensions in the software from one extension to another simultaneously.

To change multiple extensions simultaneously:

- 1 Type **change extension-station *n***, where *n* is the extension number that you want to change. Press **Enter**.

The system displays the **Change Station Extension** screen ([Figure 206, Change Station Extension screen](#), on page 800).



CAUTION:

The **Change Station Extension** screen does not change the emergency location extension that is administered within the system. You cannot use this screen to change an extension that is administered as an emergency location extension on either the **Station** screen or the **IP Address Mapping** screen.

Figure 206: Change Station Extension screen

change extension-station xxxxxxxx		Page 1 of 1
CHANGE STATION EXTENSION		
Station Name: xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx		Port: xxxxxx
FROM EXTENSION		TO EXTENSION
-----		-----
Station: xxx-xxxx		xxx-xxxx
Message Lamp: xxx-xxxx		xxx-xxxx
Emergency Location Ext.: xxx-xxxx		xxx-xxxx
IP Parameter Emergency Location: xxx-xxxx		See IP-Network Map Form

- 2 In the TO EXTENSION fields:
 - a Type a new extension that you want the current extension changed to.
 - b Type a new extension for the message lamp extension.
 - c Type a new extension for the emergency location ext. field.
 - d In the IP Parameter Emergency Location field, the system displays the words See Ip-network Map Form.

You can change the IP Parameter Emergency Location field only on the *IP Address Mapping* screen. Use the **change ip-network-map** command.
- 3 Press **Enter** to save your changes and submit the form.

Function

When you complete and submit the *Change Station Extension* screen, all administration that was associated with the previous extension is now associated with the new extensions. This administration includes any references used in a vector, in coverage, and elsewhere in the system. Once you change an extension, the software removes all references to the previous extension.

If you change an extension that is also administered on an adjunct, you must also change the extension on the adjunct. Examples of adjuncts include voice mail and an Adjunct-Switch Application Interface (ASAI) link.

Exceptions

- The **change extension-station n** command does not change the administration that is associated with call forwarding digits and abbreviated dialing buttons on the current extension.
- To change a forwarded extension that is administered as a button, do not use the **change extension-station n** command. The extension for the call forwarded button is stored as digits rather than as a UID. Avaya recommends that you use the **list usage** command before you change any extensions.
- The **change extension-station n** command does not update the ISDN BRI SPID field for BRI telephones. To update this field, you must make manual changes on both the switch and the telephone.

Audits

- If you try to change an extension that is administered on the same switch as an emergency location extension on the *Station* screen or on the *IP Address Mapping* screen, the system displays the following warning message:

Extension exists as Emergency Location. Continue?

Click **yes** to process the change. Click **no** to stop the process.
- The **change extension-station n** command is be denied under the following conditions:
 - If the extension being changed is active on a call
 - If the administrator is accessing the extension
- If you attempt to change an extension that is administered on the same system as a media complex extension on the *Station* screen, the extension cannot be changed to a 6-digit or 7-digit number. The media complex extension field on the *Station* screen does not support 6-digit or 7-digit numbers.

For example, extension 50002 is an H.323 telephone extension. Extension 50002 is administered on extension 51234 as the media complex. You cannot use the **change extension-station n** command to change extension 50002 to 7050002 because the software does not support a 7-digit media complex number. You can, however, use the command to change 50002 to 52222, because 52222 is a 5-digit extension.

Prepending numbers to the dialed string

The software allows a new value in the Insert Digits field in the **Uniform Dial Plan Table** screen. The value is **Lx**. In the value **Lx**, **x** indicates a number from 1 to 5. The number **x** must match the number of digits in the Prefix field on the **Locations** screen. For example, if the Prefix field on the **Locations** screen contains three digits, assign the value **L3**.

When you use the value **Lx** in the **Uniform Dial Plan Table** screen, the software takes the value in the Prefix field from the **Locations** screen, and prepends the number of digits to the dialed string.

To prepend numbers to the dialed string:

- 1 Type **uniform-dialplan n**, where **n** is the number of the dial plan. Press **Enter**.

The system displays the **Uniform Dial Plan Table** screen ([Figure 207, Uniform Dial Plan Table screen](#), on page 802).

Figure 207: Uniform Dial Plan Table screen

change uniform-dialplan 3

Page 1 of 2

UNIFORM DIAL PLAN TABLE

Percent Full: 0

Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num	Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
3	3	0	L3	ext	y							n	
5	5	1	005	aar	y							n	
538	7	0		aar	y							n	
6	5	1	006	aar	y							n	
852	7	0		aar	y							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	

- 2 In the Insert Digits field, type **L** and a number from 1 to 5. The number that you type must equal the number of digits in the Prefix field on the **Locations** screen. This number is the number of digits that the software prepends to the dialed telephone number.

For more information on the **Uniform Dial Plan Table** screen, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

- 3 Press **Enter** to save your changes.

Setting up announcements

In addition to the shortened extensions, you must also administer announcements. If your organization has multiple locations with announcements in multiple languages, you must administer the announcement for each location, even if many of the announcements are identical.

For example, if your organization has 100 locations, announcements in 90 locations are in English. In the remaining 10 locations, announcements are in non-English languages. You must administer announcements in all 100 locations, even if 90 of the locations use the same English announcements.

For some announcements, such as the DID/Tie/Intercept announcement, you can administer only a single announcement. This restriction can create a problem in a single distributed network. For example, an S8700 Media Server might support multiple countries, and therefore require that announcements play in the language of the country. This example poses a problem because you can only administer a single announcement.

To accommodate several announcements that share a single administered field, use the Multi-Location Dial Plan feature. With the Multi-Location Dial Plan feature, the announcement that plays is based upon the calling number, and the `Location Prefix` field on the **Locations** screen.

For example, 4567 is the extension administered in the DID/Tie/ISDN Intercept Announcement field on the **System Features** screen. You can record multiple announcements for extension xxx-4567:

- 420-4567 in English
- 813-4567 in German
- 964-4567 in French
- 371-4567 in Spanish
- 628-4567 in a user-defined language

You must administer all announcements — 420-4567 in English, 813-4567 in German, 964-4567 in French, 371-4567 in Spanish, and 628-4567 in the user-defined language. When a person calls and is about to hear announcement 4567, the system inserts the prefix digits based on the location of the caller. The system then plays announcement 4567 in the proper language.

Setting up a local centralized answering point

With the Multi-Location Dial Plan feature, you can use vector routing or hunt groups to set up a local centralized answering point (LCAP). The LCAP gives users access to a local attendant.

With vector routing, you can administer a VDN with extension “0” with a vector “route-to” step for a shorter extension. You then administer the **Uniform Dialing Plan** screen with an entry for the shorter extension in the vector step. The purpose is to prepend digits from the calling party number. The system routes the call to the extension that you designate as the LCAP.

You can also use a hunt group to set up an LCAP. You administer the **Uniform Dialing Plan** screen so that the shorter extension for the hunt group prepends digits for the calling party number. One of the members of a local hunt group can be designated as the LCAP.

Note that if you use an LCAP, you cannot also use local access codes. LCAPs and multiple feature access codes for the attendant are mutually exclusive.

Setting up multiple feature access codes for attendants

Without the Multi-Location Dial Plan feature, you can administer only one attendant feature access code (FAC) on the *Dial Plan Analysis Table* screen. With this feature, you can administer multiple attendant FACs. The Multi-Location Dial Plan feature adds the administration of the attendant dial access code to the *Feature Access Code (FAC)* screen. The `call_type` is `attnd`. This change allows different locations to share the value that is used for the attendant access.

For example, you might enter the access code **9** for ARS on the *Feature Access Code (FAC)* screen. You might then enter the attendant access code on a *Locations* screen as **8**. When a user dials **8** from that location, the software routes the call to an attendant.

Only one attendant group can exist in the software at a time.

Setting up multiple feature access codes for ARS

Without the Multi-Location Dial Plan feature, you can administer only two attendant FACs on the *Feature Access Code (FAC)* screen. With this feature, you can administer multiple ARS FACs. You can either use the global ARS codes, or provide ARS codes for a location.

For example, you might enter the access code **9** for ARS on the *Feature Access Code (FAC)* screen. You might then enter the ARS access code on a *Locations* screen as **1**. When a user dials **1** from that location, the system routes the call with ARS. A user at that location cannot dial **9** to reach ARS.

Reports for Multi-Location Dial Plan

The following reports provide information about the Multi-Location Dial Plan feature:

- None

Considerations for Multi-Location Dial Plan

This section provides information about how the Multi-Location Dial Plan feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the Multi-Location Dial Plan under all conditions.

The following considerations apply to the Multi-Location Dial Plan feature:

- None

Interactions for Multi-Location Dial Plan

This section provides information about how the Multi-Location Dial Plan feature interacts with other features in your system. Use this information to ensure that you receive the maximum benefits of the Multi-Location Dial Plan in any feature configuration.

- Attendant

This feature allows you to administer multiple attendant codes. This feature does not allow you to administer multiple attendant groups. Only one attendant group is allowed, unless you have Attendant Partitioning. This feature provides a way to support multiple local centralized answering points (LCAPs). The LCAPs do not use attendant groups.

- Attendant Vectoring

Attendant Vectoring, if enabled, takes precedence over any local attendant codes that are administered. The software uses call vectors to process calls to an attendant. Such calls are:

- Local attendant codes
- The attendant code on the *Dial Plan Analysis Table* screen
- The attendant access code on the *Feature Access Code (FAC)* screen

- Automatic Circuit Assurance (ACA)

The ACA Referral Destination field on the *System Features Parameters* screen requires that an attendant group exists. This field also requires either:

- The **attdd** administration on the *Dial Plan Analysis Table* screen, or
- Administration of the attendant access code on the *Feature Access Code (FAC)* screen

- Automatic Wakeup

If a telephone has Automatic Wakeup requests pending when you run the **change extension-station** command, the system cancels the wakeup requests.

- Call Forwarding

Any call forwarding information that is stored with an extension is lost when you use the **change extension-station** command. If the telephone that you change with the **change extension-station** command is a forwarded-to telephone, you must manually update the extension that uses the forwarded-to extension. If you do not manually update the extension, the system does not forward calls correctly.

- Call Park

This feature does not support common shared extensions to park calls. Since common shared extensions are not assigned to physical telephones, the range of common shared extensions can be shared in all locations.

- Crisis Alert

The **change extension-station** command does not update the Originating Extension field on the *Crisis Alert System Parameters* screen. You must manually update the Originating Extension field if you change the originating extension field.

- Leave Word Calling (LWC)

The Stations with System-wide Retrieval Permission for the Leave Word Calling Parameters field on the *System Features Parameters* screen requires that an attendant group exists. This field also requires either:

- The **attd** administration on The *Dial Plan Analysis Table* screen, or
- Administration of the Attendant Access Code on the *Feature Access Code (FAC)* screen

- Local Spare Processor (LSP)

If you run the **change extension-station** command on the controller, but do not also save the translations to the LSP, the two system translations might not be synchronized.

Use either of these following commands to save the translations to the LSP:

- The **save trans lsp** command locally saves the translations, and performs a filesync operation to all registered LSPs.
- The **save trans lsp <IP address>** command, where **<IP address>** is the IP address of a specific LSP, locally saves the translations, and performs a filesync operation to the specified LSP.

- Night Service

This feature does not support location-based Night Service. Therefore, you might want to restrict calls to attendants who are local to the calling party. The attendant most likely speaks the same language as the caller. At night, instead of putting the entire system into Night Service, you might want an attendant to put only one location into Night Service.

To accomplish this, you can use hunt groups as attendant queues. Each hunt group can be put into Night Service separately, and have its own Night Service destination. You can also administer the Night Service destination by tenant, trunk group, and trunk group number.

- Security Violations Notification

The referral destination fields on the *Security-Related System Parameters* screen requires that an attendant group exists. This field also requires either:

- The **attd** administration on the *Dial Plan Analysis Table* screen, or
- Administration of the attendant access code on the *Feature Access Code (FAC)* screen

The referral destination fields are:

- SVN Login Violation Notification Enabled
- SVN Remote Access Violation Notification Enabled
- SVN Authorization Code Violation Notification Enabled
- SVN Station Security Code Violation Enabled

- Station Hunting

Changing a telephone extension with the **change extension-station** command maintains the hunting chain.

- Survivable Remote EPN/WAN Spare Processor

If you run the **change extension-station** command in a configuration where a survivable remote processor exists, you must also run the command on the survivable processor. If you do not, the extensions that you changed on the server do not exist on the survivable remote when the remote processor takes control.

- Uniform dial plan (UDP)

The **change extension-station** command does not update extensions that are in the *Uniform Dial Plan Table* screen. Avaya supports external system management tools that handles changes to the UPD table.

Multifrequency Signaling

Use the Multifrequency (MF) signaling feature to perform signaling used between media servers 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 usually a combination of two frequencies from a group of 5 or 6 frequencies (2/5 or 2/6). The origination and destination servers or switches exchange tones that have specific meanings according to the MF protocol.

Detailed description of Multifrequency Signaling

This section provides a detailed description of the Multifrequency Signaling feature.

Avaya Communication Manager 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 media server over analog or digital CO, Direct Inward Dialing (DID), or Direct Inward and Outward Dialing (DIOD) trunks. It also provides signaling between any 2 servers running Communication Manager.

Communication Manager provides MF signaling that complies with ITU regulations and national regulations for specific countries. It provides these types of MF signaling: MFE MF Shuttle, and multifrequency compelled (R2-MFC). These protocols signal the called number, the calling party's number (ANI), and information about the type of call or type of caller (category).

Communication Manager allows prefix digits for ANI sent on outgoing calls to be defined per server, or per the originator's class of restriction.

If a call is a tandem call and the incoming and outgoing trunk use different protocols, Communication Manager makes no attempt to convert between the various protocol's meanings for category. Instead,

- the server uses the incoming trunk's COR assigned category if the outgoing trunk is Russian or R2-MFC, and
- the server uses ARS call types if the outgoing trunk is MFE.

The server running Communication Manager provides the incoming ANI to all features on Communication Manager 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 types 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, 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 multifrequency compelled (R2-MFC) signaling allows each country to define the meanings of the R2 frequency combinations.

Guidelines for administering Multifrequency Signaling

To administer MF signaling, you must first identify the origination and destination server or media servers. The one making the call is the origination server. The one answering the call is the destination server.

- The origination server/media server creates *forward* signals, classified as group I and group II signals.
- The destination server/media server 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 server (forward group I and group II signals) and destination server (backward group A and group B signals).

Forward		Backward
Group I	digit -->	
	<-- A.1	Group A
	digit -->	
	<-- A.1	

Forward		Backward
	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 each 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 might differ by country) assigned to the identifying codes are the same. However, the messages assigned to the identifying codes can 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 the **AAR and ARS Digit Analysis Table** screen, set the ANI Req field to **y**, or on the **AAR and ARS Digit Conversion Table** screen, set the ANI Req field to **y**.

Hardware requirements for Multifrequency Signaling

The Multifrequency Signaling feature requires the following hardware:

- None

Administering Multifrequency Signaling

The following steps are part of the administration process for the Multifrequency Signaling feature:

- None

Screens for administering Multifrequency Signaling

Screen name	Purpose	Fields
<i>Multifrequency-Signaling Related Parameters</i>	Sets the system parameters for multifrequency signaling	All
<i>AAR and ARS Digit Conversion Table</i>	Sets automatic number identification	ANI Req'd
<i>AAR and ARS Digit Analysis Table</i>	Sets automatic number identification	ANI Req'd

Reports for Multifrequency Signaling

The following reports provide information about the Multifrequency Signaling feature:

- None

Considerations for Multifrequency Signaling

This section provides information about how the Multifrequency Signaling feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Multifrequency Signaling under all conditions. The following considerations apply to Multifrequency Signaling.

The following considerations apply to R2-MFC only:

- 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 displays, ANI information and the CDR records the information.
- When Avaya Communication Manager 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 Communication Manager.
- If Avaya Communication Manager makes an outgoing call to the CO that uses an open numbering plan, the CO must send the signal A.1 to Communication Manager after sending the last address digit to the CO. Then, the CO should time out and send a pulsed signal A.3 to Communication Manager requesting the Group II signal.

- Avaya Communication Manager offers the option to record the Calling Party Category in the CDR. For incoming external calls, this comes from the Group II signal. For internal calls and station-originated external calls, this comes from the 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 is capable of receiving this information.
- You can assign Calling Party Category and Called Party Category on a trunk-by-trunk basis.
- You can record an announcement to play when outgoing R2-MFC trunk calls do not complete. This applies when Communication Manager receives either group A or B signals from the called Central Office or other media server or switch.

Interactions for Multifrequency Signaling

This section provides information about how the Multifrequency Signaling feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Multifrequency Signaling in any feature configuration.

- **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 COR checking, ANI prefix and ANI truncation still apply.
- **Attendant Console**
If the attendant assists or extends a call for a station using 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 applies to the primary extension number of bridged call appearances.
- **Call Detail Recording (CDR)**
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 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.

NOTE:

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.

NOTE:

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 server 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.

NOTE:

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.

Multiple Level Precedence and Preemption

Use the Multiple Level Precedence and Preemption (MLPP) feature to allow users to request priority processing of their calls during critical situations.



CAUTION:

MLPP is currently designed to meet only Generic Switching Center Requirements (GSCR) for connection to a Defense Switched Network (DSN) by federal, state, or local government agencies. MLPP is not currently designed for use in commercial enterprise environments. Activation of this feature in any other type of network environment can result in unexpected or unwanted feature operations.

Detailed description of Multiple Level Precedence and Preemption

This section provides a detailed description of the Multiple Level Precedence and Preemption feature.

The Multiple Level Precedence and Preemption feature is available with Avaya Communication Manager release 2.0 (V12) or later.

While the media servers and media gateways that are referenced in this document support MLPP, the Joint Interoperability Test Command (JITC) has only certified the following servers:

- DEFINITY Server CSI
- DEFINITY Server SI
- Avaya S8700 Media Server supporting an MCC1 or SCC1 Media Gateway

Multiple Level Precedence and Preemption supports the following capabilities:

- Precedence Calling
- Announcements for Precedence Calling
- Precedence Call Waiting
- Precedence Routing
- Dual Homing
- End Office Access Line Hunting
- Preemption
- Line Load Control
- Worldwide Numbering and Dialing Plan (WNDP)

Precedence Calling

Precedence Calling allows users, on a call-by-call basis, to select a level of priority for each call. The need of the user and the importance of the call is the basis for the priority. The call can receive a priority routing whether the call is local, or routed around the world.

When placing calls, users can access five levels of precedence:

- Flash Override (the highest precedence level)
- Flash
- Immediate
- Priority
- Routine (the default, and lowest precedence level)

Calls dialed without specifying a precedence level are treated as Routine level precedence calls.

The administrator assigns a maximum precedence level to each telephone user. The more important or higher in rank of the user, the higher the precedence level. Users cannot originate calls at precedence levels that are higher than the maximum administered level. Non-MLPP calls are treated as routine level precedence calls.

For example, General Davis has a maximum precedence level of Flash assigned to his telephone. Without intervention, everyday calls are treated at the Routine level. One day, a crisis occurs at a military installation, and he must make an emergency call to his field commanders over the DSN. General Davis can use the Precedence Calling feature to raise the level of his call to Priority, Immediate, or Flash. When he places this call, the communication server gives the call priority handling, and the call is sent over the DSN access line.

The format for Precedence Calling dialed digits is as follows:

Access digits		Address digits		
FAC	Precedence digit	Area code	Office code	Extension
A	P	[NXX]	NXX	XXXX

Where:

A is the Precedence Calling FAC

P is any digit 0–4 (digits 5–9 can also be used)

X is any digit 0–9

N is any digit 2–9

[] indicates optional digits

- Access digits – The access digits are comprised of the Precedence Calling feature access code (FAC) followed by a Precedence digit. The single-digit code used for the Precedence digit is administered as shown on [Assigning Precedence Calling system parameters](#) on page 834.

The default precedence level digits are:

- 0 - Flash Override
 - 1 - Flash
 - 2 - Immediate
 - 3 - Priority
 - 4 - Routine
- Address digits – The address digits are the seven-digit or ten-digit telephone number.

Precedence calls above the Routine level use special precedence ringback tones for the calling party, and special ringing pattern for the called party. The TN2182B or TN2182C Tone Clock circuit pack generates the Precedence calling tones on the following media gateways and servers:

Media Gateways	Supported by servers
• G650 Media Gateway • SCC1 Media Gateway • CMC1 Media Gateway • MCC1 Media Gateway	• DEFINITY Server • S8500 Media Server • S8700 Media Server

On a G350 or G700 Media Gateway supported by an S8300, S8500, or S8700 Media Server, the Media Gateway processor generates precedence calling tones. The S8100 Media Server processor also generates tones.

- The ringback tone is a 1.65 second burst of mixed 440 Hz and 480 Hz tone, followed by 0.35 seconds of silence. This tone repeats until the call is answered, the caller hangs up, or until the Precedence Call Timeout occurs (see [Assigning Precedence Calling system parameters](#) on page 834 for more information).
- The ringing pattern for precedence calls is the same pattern used with Priority Calling, which is a 3-burst ring.

Precedence Calling diversion scenarios

When a precedence call to a telephone goes unanswered, the system attempts to connect the caller to a backup point as follows:

- 1 The call is diverted to the attendant console.
- 2 If the console is in Night Service or there is no console administered, the call is diverted to a night station.
- 3 If there is no console or night station administered, the call can be diverted to a user-defined telephone (called the Remote Attendant Route String).

The Attendant Diversion Timing controls how long this type of call rings before the call is routed to the Remote Attendant Route String. The Remote Attendant Route String is any valid telephone number on the network. Usually a backup answering position for the remote attendant console. The Remote Attendant Route String does not raise the precedence level of the call.

- 4 If the Remote Attendant Route String is not defined and there is no attendant console or night station, the call rings until answered or abandoned.

This precedence call diversion scenario has variations for DSN calls, non-DSN calls, and local calls:

- DSN Calls – If an outgoing precedence call over a DSN trunk is not answered after an administrable period of time, the call routes to:
 - the attendant console or night station on the remote communication server, or
 - a user-defined telephone (the Remote Attendant Route String)

The administrable period of time is the Precedence Call Timeout on the remote communication server.

- Non-DSN Calls – If an outgoing precedence call over a non-DSN trunk is not answered after an administrable period of time, the call routes to:
 - a local attendant console or night station on the local communication server, or
 - a user-defined telephone (the Remote Attendant Route String).

The administrable period of time is the Attendant Diversion Timing on the local communication server.

- If a local, intraswitch precedence call is not answered after an administrable period of time (the Precedence Call Timeout), the call routes to:
 - a local attendant console
 - to a night station, or
 - a user-defined telephone (the Remote Attendant Route String)

NOTE:

For a precedence call that diverts to a night station or to a Remote Attendant Route String, the number must be administered in the Precedence Routing digit-conversion tables. For more information, see [Assigning digit conversion](#) on page 847 and [Assigning Precedence Calling system parameters](#) on page 834.

When calls are redirected, a Call Purpose Indicator is displayed on the attendant console and on display telephone sets to indicate the precedence level of the call. The following indicators are provided:

- FO - Flash Override
- FL - Flash
- IM - Immediate
- PR - Priority

Routine precedence calls do not have a Call Purpose Indicator.

When callers attempt to use a precedence level higher than authorized, the caller hears the “Unauthorized precedence level attempted” recording. If an announcement is not assigned, the caller hears intercept tone.

The following table shows how precedence calls are processed depending on the precedence level and the administered maximum precedence level of the caller:

Maximum precedence level of the user	Precedence level of call	Call treatment
Flash Override	Flash Override	Call completes normally
Flash Override	Flash	Call completes normally
Flash Override	Immediate	Call completes normally
Flash Override	Priority	Call completes normally
Flash Override	Routine	Call completes normally
Flash	Flash Override	Recorded announcement or intercept tone
Flash	Flash	Call completes normally
Flash	Immediate	Call completes normally
Flash	Priority	Call completes normally
Flash	Routine	Call completes normally
Immediate	Flash Override	Recorded announcement or intercept tone
Immediate	Flash	Recorded announcement or intercept tone
Immediate	Immediate	Call completes normally
Immediate	Priority	Call completes normally
Immediate	Routine	Call completes normally
Priority	Flash Override	Recorded announcement or intercept tone
Priority	Flash	Recorded announcement or intercept tone
Priority	Immediate	Recorded announcement or intercept tone
Priority	Priority	Call completes normally
Priority	Routine	Call completes normally
Routine	Flash Override	Recorded announcement or intercept tone
Routine	Flash	Recorded announcement or intercept tone
Routine	Immediate	Recorded announcement or intercept tone
Routine	Priority	Recorded announcement or intercept tone
Routine	Routine	Call completes normally

How service domains influence preemption and precedence

You define MLPP service domains at two levels:

- in the ***Class of Restriction*** screen, where you then assign the COR to a telephone of a user
- in the ***Multiple Level Precedence & Preemption Parameters*** screen, which applies to all resources on the server

Multiple Level Precedence and Preemption

Detailed description of Multiple Level Precedence and Preemption

The system uses the service domain to determine whether the system applies the preemption or precedence level of the caller to potentially preempt a routine call or another call of lower precedence. If an existing call has a different service domain than the service domain of the call attempting a preemption, preemption is not allowed, regardless of the preemption levels of the two calls.

The two levels of service domain influence preemption and precedence capabilities in different ways, depending on which of the following resources the server recognizes as carrying the call:

- Intraswitch routing to another telephone on the same server
- Incoming ISDN-PRI trunks for a call originating from another server on the network
- Incoming non-ISDN-PRI trunks for a call originating from another server on the network

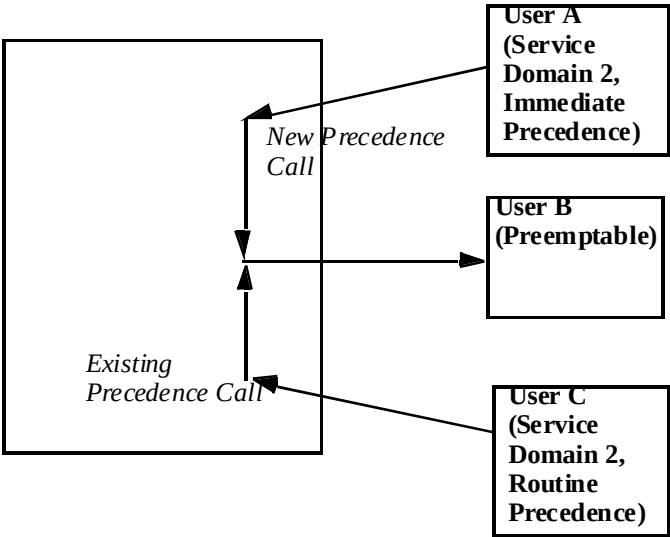
Station-to-station calls on the same server

If	Then
• a user makes an MLPP call to another user whose telephone resides on the same server, and • the called telephone is busy with an existing call	the server uses the following process to determine whether to grant precedence to the new call:

- 1 The server assigns the existing call to the service domain defined in the COR of the caller.
- 2 The server checks the COR of the new caller to define the service domain of the new call.
- 3 The server matches the service domain of the new call with the service domain of the existing call.
- 4 If the service domains match, then the server matches the precedence level of the existing call with the precedence level of the new call. If the new call has a higher level, the new call preempts the existing call.

If the service domains do not match, then the server gives the new call an announcement that says the MLPP call cannot be completed. See [Figure 208, Station-to-station calls on the same server](#), on page 821.

Figure 208: Station-to-station calls on the same server



In [Figure 208](#), the server uses the service domains and precedence levels to treat calls as follows:

- 1 User C has made a routine call to User B and is still connected. The server has checked the service domain of the COR of User C, and assigned service domain 2 to the call.
- 2 User A makes an immediate precedence call to User B. The server checks the service domain of the COR of User A, and assigns service domain 2 to the call.
- 3 The server matches the service domain of User C with the service domain of User A.
- 4 Because the calls are in the same service domain, and because User A has used a higher precedence level than User C, the server allows User A to preempt the call of User C.

Precedence calls to destinations over ISDN-PRI trunks

If	Then
• a user makes an MLPP call to another user whose telephone resides on a different MLPP server, and • the call <i>arrives</i> at the server of the called user over an ISDN-PRI trunk, and • the called telephone is busy with an existing call	the destination MLPP server uses the following process to determine whether to grant precedence to the new call:

- 1 The server assigns the existing call to the service domain defined either in the COR of the caller, or by its own system service domain.
- 2 The server checks the COR of the new caller to define the service domain of the new call.

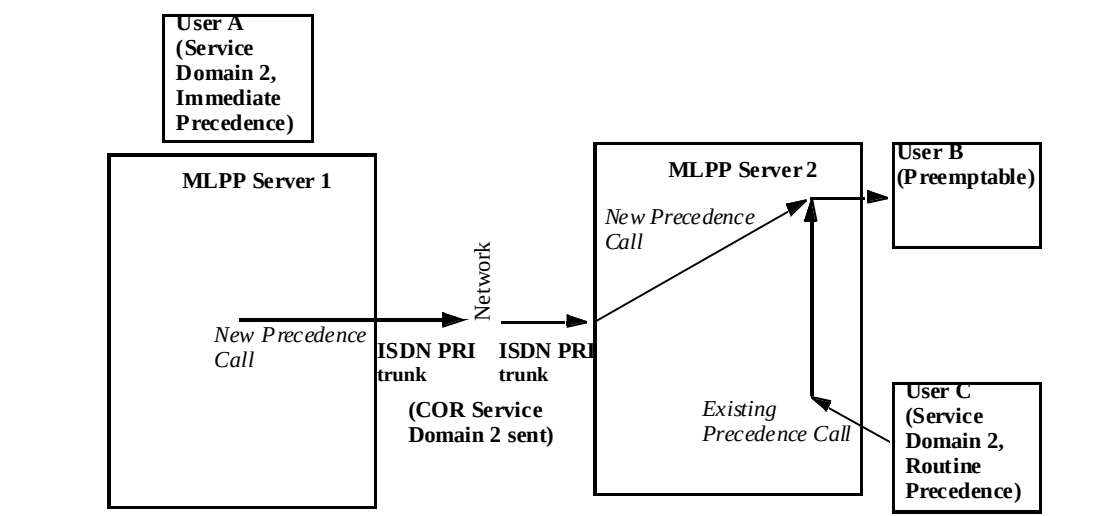
Multiple Level Precedence and Preemption

Detailed description of Multiple Level Precedence and Preemption

- 3 The server matches the service domain of the new call with the service domain of the existing call.
- 4 If the service domains match, then the server matches the precedence level of the existing call with the precedence level of the new call. If the new call has a higher level, the new call preempts the existing call.

If the service domains do not match, then the server gives the new call an announcement that says the MLPP call cannot be completed. See [Figure 209, Precedence calls to destinations over ISDN-PRI trunks](#), on page 822.

Figure 209: Precedence calls to destinations over ISDN-PRI trunks



In [Figure 209](#), the destination server, Server 2, uses the service domains and precedence levels to treat calls as follows:

- 1 User C has made a routine call to User B and is still connected. The server has checked the service domain of User C's COR and assigned service domain 2 to the call.
- 2 User A, who is on Server 1, makes a precedence call to User B. Server 2 checks the service domain on the incoming ISDN call and finds service domain 2 (defined in User A's COR on Server 1). Server 2 assigns service domain 2 to the call.
- 3 The server matches the service domain of User C with the service domain of User A.
- 4 Because the calls are in the same service domain, and because User A has used a higher precedence level than User C, the server allows User A to preempt User C.

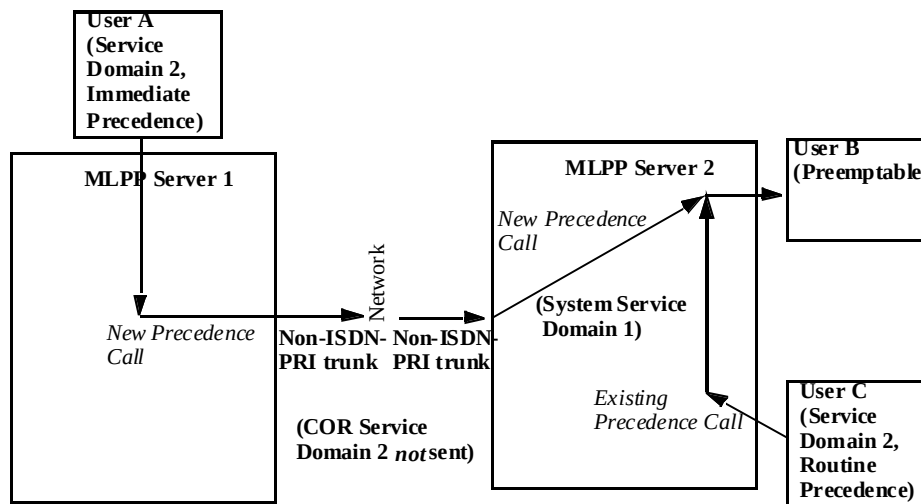
Precedence calls to destinations over non-ISDN-PRI trunks

If	Then
- a user makes an MLPP call to another user whose telephone resides on a different MLPP server, and - the call <i>arrives</i> at the called user's server over a non-ISDN-PRI trunk, and - the called telephone is busy with an existing call	the destination server uses the following process to determine whether to grant precedence to the new call:

- 1 The server assigns the existing call to the service domain defined either in the caller's COR or by its own system service domain.
- 2 The server checks its own system service domain to define the service domain of the new call.
- 3 The server matches the service domain of the new call with the service domain of the existing call.
- 4 If the service domains match, then the server matches the precedence level of the existing call with the precedence level of the new call. If the new call has a higher level, the new call preempts the existing call.

If the service domains do not match, then the server gives the new call an announcement that says the MLPP call cannot be completed. See [Figure 210, Precedence calls to destinations over non-ISDN-PRI trunks](#), on page 823.

Figure 210: Precedence calls to destinations over non-ISDN-PRI trunks



In [Figure 210, Precedence calls to destinations over non-ISDN-PRI trunks](#), on page 823, the system uses the service domains and precedence levels to treat calls as follows:

- 1 User C has made a routine call to User B and is still connected. The server has checked the service domain of the COR of User C and assigned service domain 2 to the call.
- 2 User A, who is on Server 1, makes a precedence call to User B. Because the call from User A arrives at Server 2 on a non-ISDN-PRI trunk, Server 2 does not receive the service domain identifier of User A. Server 2 assigns the system service domain 1 to the call.
- 3 The server matches the service domain of User C with the service domain of User A.

Because the calls of users A and C are *not* in the same service domain, calls from User A receive an announcement that says the MLPP call cannot be completed.

Announcements for Precedence Calling

In certain situations, precedence calls are blocked because of unavailable resources or improper use. The MLPP feature requires special announcements to notify users when Precedence Calling (for calls higher than Routine precedence) is denied, or when service is not available.

The announcements that MLPP uses include:

- Blocked precedence call
This announcement plays when the system attempts to preempt an existing call with a call that has a precedence level higher than Routine precedence that is also equal to or lower than the precedence level of the current call. If an announcement extension is not assigned, the caller hears reorder tone (fast busy).
- Unauthorized precedence level attempted
This announcement plays when a caller attempts to place a precedence call with a precedence level that is higher than authorized by their Class of Restriction (COR). If an announcement extension is not assigned, the caller hears intercept tone (siren tone).
- Service interruption prevented call completion
This announcement plays when a service interruption prevents a precedence call from being completed. If an announcement extension is not assigned, the caller hears reorder tone (fast busy).
- Busy, not equipped for Preemption or Precedence Call Waiting
This announcement plays when a precedence call is placed to a busy line and the line does not have Precedence Call Waiting or is not preemptable. If an announcement extension is not assigned, the caller hears reorder tone (fast busy).
- Vacant code
This announcement plays when a precedence call is placed to an unassigned extension. If an announcement extension is not assigned, the caller hears reorder tone (fast busy).

If a caller is using Routine precedence and the call cannot be completed for any of these reasons, the caller hears busy tone.

Precedence Call Waiting

After the system routes a precedence call, the called party might already be busy on another call. Precedence Call Waiting allows the caller to “camp on” to the line of the called party, and wait for the called party to answer the call. The caller hears a special ringback tone, and the called party hears a call waiting tone. Depending on the type of telephone being used, the called party:

- can put the current call on hold and answer the incoming call, or
- must hang up on the current call to answer the incoming call

Precedence Routing

When precedence calls are destined for other switches in a network, the Precedence Routing feature routes the calls. The Precedence Routing feature routes calls based on three main criteria:

- the destination number
- the precedence level
- the time of day

These routing criteria are administrable and can be changed as required. Two related features are Dual Homing and End Office Access Line Hunting.

Dual Homing

Dual Homing allows a user to dial a telephone number and have the call route to its destination over alternate facilities if the initial route is unavailable. This operation is transparent to the user and no special dialing is required.

Dual Homing uses the Precedence Routing feature to provide alternate routing to nodes on a DSN. If a call fails to complete over the first trunk access line, the call is rerouted over a different trunk access line. This process can continue for any number of alternate routes.

If the call fails to complete by the time it gets to the last trunk access line, the system routes the call either to:

- Busy tone, or
- The “Blocked precedence call” recorded announcement (see [Announcements for Precedence Calling](#) on page 824).

For example, a user dials a DSN number, such as 345-8854. Using Precedence Routing, you administer all calls beginning with the digits “345” to rout as follows:

- first route over trunk group 20
- then route over trunk group 21
- finally route over trunk group 22.

If all trunks in trunk group 20 are busy, the system next checks for idle trunks in trunk group 21, and finally in trunk group 22. If all trunks in all three trunk groups are busy, the call routes to fast busy tone or to a recorded announcement.

For a more detailed description of the available routing options, see [Precedence Routing](#) on page 825.

End Office Access Line Hunting

The End Office Access Line Hunting feature automatically hunts for an idle trunk over end office access lines. This feature hunts for an idle trunk based on the precedence level of the call. The search occurs over either a preemptable trunk group or a nonpreemptable trunk group.

For calls higher than Routine precedence, the system hunts for an idle trunk in a preemptable trunk group. The following steps detail the hunting algorithm:

- 1** If the system finds an idle trunk, the system provides precedence ringing.
- 2** If the system does not find an idle trunk, the system reexamines a preemptable trunk group on a preemptive search. The system preempts an active call of the lowest available precedence level.
- 3** The system hunts for an idle trunk in a nonpreemptable trunk group. If the system finds an idle trunk, the system provides precedence ringing.
- 4** If the system cannot find a trunk, the call is routed to the "Blocked precedence call" recorded announcement (see [Announcements for Precedence Calling](#) on page 824). If announcements are not recorded or administered, the caller hears reorder tone.

For Routine precedence calls, the system hunts for an idle trunk in a nonpreemptable trunk group and attempts to connect the call. If the system does not find an available trunk, the caller hears busy tone. For more information about preemptable and nonpreemptable trunks, see [Preemption](#) on page 826.

Preemption

Preemption works with Precedence Routing to further extend the call routing capabilities of MLPP. Preemption actually disconnects an existing, lower-priority call to complete a more important precedence call. Even non-MLPP calls are treated as routine level precedence calls, and can be preempted.

When preemption occurs, the callers on the existing call hear a tone indicating that the call is about to be preempted. The callers have three seconds to end the call before the call is automatically disconnected. After the existing call is disconnected, the new call is placed using preempted facility.

Line Load Control

Line Load Control (LLC) is a feature that restricts a predefined set of telephone users from originating calls during a crisis or emergency. This is done by systematically reducing the number of telephones that can originate calls during high-traffic periods. This situation is sometimes called a "lockdown." When the lockdown situation passes, the LLC restriction levels can be reduced or removed completely.

Users are assigned to a Line Load Control level based on their relative importance. When an emergency occurs, the administrator manually activates the feature to restrict calls by users of lower importance. When the emergency is over, the administrator manually disables the feature.

For example, if a security emergency occurs, telephone users who are responsible for managing the crisis are not restricted from originating calls, but other telephone users, for example, in the accounting department, are restricted. When the crisis is over, the administrator returns the system to normal operation.

There are four levels at which this feature can be controlled. These system levels determine what telephones, based on the Class of Restriction (COR), are restricted from originating calls. This feature does not restrict incoming calls, or calls originating from an attendant console or a night station.

The system levels are as follows:

- 0 – (default) Feature is not active. There are no restrictions.
- 2 – Restrict telephones with a COR assigned to LLC levels 2, 3, and 4
- 3 – Restrict telephones with a COR assigned to LLC levels 3 and 4
- 4 – Restrict telephones with a COR assigned to LLC level 4.

System level 1 is not a valid value. The LLC feature cannot restrict telephones with a COR assigned to LLC level 1 from originating calls.

When LLC is activated, the system restricts all telephones with a COR at that LLC level and below from originating any calls. If a restricted telephone is already active on a call when the restriction is activated, the call is not interrupted or disconnected. The telephone becomes restricted only after the user hangs up from the active call.

When the need for LLC has passed, the administrator can change the LLC to a less-restrictive level or completely deactivate the feature.

Using the following table, this is how the LLC feature can be used. For this example, the LLC is at Level 0 (no restrictions), and telephone 2635 is active on a call.

Telephone	COR	LLC Level
5300	11	1
5350	12	2
2540	13	3
3300	14	4
2635	14	4

- 1** Because of high telephone traffic, the system administrator changes the LLC to level 3.
Telephones 2635, 2540, and 3300 cannot originate calls because the assigned COR LLC level for these telephones is equal to or less than the system LLC level.
The active call on telephone 2635 is not disturbed. As soon as telephone 2635 hangs up, that telephone cannot originate calls.
Telephones 5300 and 5350 can originate calls because their assigned COR LLC level is higher than the system LLC level.
- 2** Call traffic is still too high, so the system administrator changes the LLC to level 2.
Now telephone 5350, in addition to telephones 2635, 2540, and 3300, cannot originate new calls. Telephone 5300 can still originate new calls.
- 3** Call traffic subsides. The system administrator changes the LLC back to level 0.
All telephones can now originate calls.

Worldwide Numbering and Dialing Plan

The Worldwide Numbering and Dialing Plan (WNDP) feature is compatible with the standard numbering system established by the Defense Communications Agency (DCA). WNDP is a dialing system used in a DSN. WNDP is similar to [Precedence Calling](#), but the pattern of digits that users dial is different.

The format of the dialed digits is as follows:

Access digits		Address digits		
FAC	Route code	Area code	Office code	Extension
A	[[1]X]	[NXX]	NXX	XXXX

Where:

A is the two-digit WNDP FAC for the precedence level

1 is a Route Code setup digit.

X is any digit 0-9

N is any digit 2-9

[] indicates optional digits

- FAC – The FAC is the set of two-digit WNDP FACs. Each precedence level uses a unique FAC.
 - Flash Override – for example, **90**
 - Flash – for example, **91**
 - Immediate – for example, **92**
 - Priority – for example, **93**
 - Routine – for example, **94**
- Route Code – After dialing the FAC, the user has the option of dialing a Route Code. The Route Code is a two-digit DSN code that consists of a Route Code Setup digit and the Route Code digit. The Route Code allows the user to inform the communication server of special routing or termination requirements. The Route Code is limited to the DSN. The Route Code determines whether a call uses data-grade or voice-grade trunking. The Route Code is also used to indicate that the dialed number is either a Federal Telephone System (FTS) or a Continental U.S. (CONUS) commercial number.

If you do not require special call features and you want to use the default value zero (0), you do not have to use the Route Code. If you want to use a value other than the default value zero (0), you must use the Route Code.

- The first digit of the Route Code is the Route Code Setup digit. The Route Code Setup digit is the number **1**. The Route Code Setup digit indicates that the next digit gives instructions to the network for specialized routing. If a Route Code is dialed, the Route Code Setup digit is deleted and the second digit is saved.
- The second digit of the Route Code is the Route Code digit. Valid entries are:
 - 0** = Voice call (the default value)
 - 1** = Circuit switched data call
 - 2** = Satellite avoidance call

- 3 = (reserved)
- 4 = (reserved)
- 5 = Hotline voice grade call
- 6 = Hotline data grade call
- 7 = (reserved)
- 8 = (reserved)
- 9 = (reserved)

The Route Code digit becomes part of the dialed number, and can be used for route selection using the Precedence Routing translations. Precedence Routing allows digit strings to be modified before outpulsing. This capability modifies the Route Code as needed by the terminating trunk group. If a Route Code digit is not dialed, the system inserts the default Route Code Digit of zero (0) as defined on the **Multiple Level Precedence & Preemption Parameters** screen. The default Route Digit routes the calls over the voice network, not the data network.

- Address Digits – The Address Digits are the seven-digit or ten-digit DSN number.

The format of the outpulsed digits is as follows:

Precedence digit	Route code	Address digits		
		Area code	Office code	Extension
P	[[1]X]	[NXX]	NXX	XXXX

The digit outpulsing is administrable using the Precedence Routing functionality (see [Precedence Routing](#) on page 825). Precedence Routing allows flexible routing of dialed numbers, and the ability to modify the digits outpulsed as needed (for example, outpulsing no Route Digit, only the Route Digit, or "1" and the Route Digit). The digits sent to Precedence Routing are of the form:

PRXXX...

Where:

- P** is the Precedence Digit
- R** is the Route Code (if WNDP is active)
- XXX...** are the Address Digits

If a particular route requires the Route Code of the form 1X, you can use the digit modification translations for the route to insert the "1." If the route does not require the Route Digit, the digit modification can be translated to delete the Route Digit. The digit modification translations insert a Default Route Digit if none is dialed.

Hardware requirements for Multiple Level Precedence and Preemption

The Multiple Level Precedence and Preemption feature requires the following hardware:

You need to use the following circuit packs to support the MLPP features. If you are installing a new system, these circuit packs are delivered with a new system. If you are upgrading an existing system, you must replace any outdated circuit packs.

Type	Number	Requirements
DEFINITY Servers, S8100 Servers, and S8500 or S8700 Media Servers using G650, SCC1, and MCC1 Media Gateways		
Analog Tie Trunk	TN760E or newer (four ports for each circuit pack)	Required for trunks that carry MLPP DTMF and MF 2/6 trunk traffic. Older version circuit packs can be used for non-MLPP traffic.
DS1 Trunk	TN464F or TN2464 Vintage 18 or newer (24 T1 channels or 32 E1 channels)	Required only for trunks that carry MLPP traffic. Older version circuit packs can be used for non-MLPP traffic. E1 trunks require a special TN464F circuit pack. See the Rapid Response Web site for more information: (http://rr-db1.dr.avaya.com/rapid_response/MLPPins2.html)
Tone Clock	TN2182B or TN2182C (eight ports for each circuit pack)	All tone clocks must be TN2182B or later, except when using MF 2/6 trunks. MF 2/6 trunks require the TN2182C, Vintage 2 or later. See the Rapid Response Web site for more information: (http://rr-db1.dr.avaya.com/rapid_response/MLPPins2.html) The S8100 Media Server does not require a tone clock circuit pack. Tone clock service is built into the processor.
IP Server Interface	TN2312AP or TN2312BP	Must be Vintage 5 or later.

The following implementations are standard:

- analog trunking with the MM711 Analog Media Module
- DS1 trunking with the MM710 E1/T1 Media Module
- announcements, built into the processor and enabled by the license file
- tone clock services, built into the processor

No special vintage media modules are required for the following servers:

- G700 or G350 Media Gateway supported by an S8300 Media Server
- S8500 Media Server
- S8700 Media Server

Administering Multiple Level Precedence and Preemption

The following steps are part of the administration process for the Multiple Level Precedence and Preemption feature:

Precedence Calling

- [Assigning an MLPP feature access code](#)
- [Assigning Precedence Calling system parameters](#)
- [Assigning a maximum precedence level](#)
- [Assigning trunks](#)
- [Assigning a hot line number](#)
- [Assigning attendant queue priorities](#)

Announcements for Precedence Calling

- [Adding extensions](#)
- [Assigning announcement types](#)
- [Recording announcements](#)
- [Deleting announcements](#)
- [Saving announcements](#)

Precedence Call Waiting

- [Enabling Precedence Call Waiting for a telephone](#)
- [Setting the Precedence Call timeout](#)
- [Assigning feature access codes](#)

Precedence Routing

- [Assigning digit analysis](#)
- [Assigning route patterns](#)

Dual Homing

- see [Precedence Routing](#)

End Office Access Line Hunting

- see [Precedence Routing](#)

Preemption

- [Assigning Preemption to a COR](#)
- [Assigning trunks for Preemption](#)
- [Setting the Precedence Call timeout](#)

Line Load Control

- [Assigning the LLC level for the system](#)
- [Assigning the LLC to a COR](#)

Worldwide Numbering and Dialing Plan (WNDP)

- [Assigning WNDP system parameters](#)
- [Assigning WNDP feature access codes](#)
- [Assigning a Hot Line number](#)
- [Assigning attendant queue priorities](#)

This section describes:

- Any prerequisites for administering the Multiple Level Precedence and Preemption feature
- The screens that you use to administer the Multiple Level Precedence and Preemption feature
- Complete administration procedures for the Multiple Level Precedence and Preemption feature

Prerequisites for administering Multiple Level Precedence and Preemption

You must complete the following actions before you can administer the Multiple Level Precedence and Preemption feature:

- On the **Optional Features** screen, ensure that the G3 Version field is set to **V12** (Communication Manager release 2.0) or later. If this field is not set to **V12** or later, your system is not enabled for the Multiple Level Precedence and Preemption feature. Contact your Avaya representative for assistance.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

- Click **Next** until you see the Multiple Level Precedence and Preemption field. Ensure that the Multiple Level Precedence and Preemption field is set to **y**. Your license file sets the value in this field. You cannot manually change this value. If the Multiple Level Precedence and Preemption field is set to **n**, see your Avaya representative for assistance.

Screens for administering Multiple Level Precedence and Preemption

Screen name	Purpose	Fields
Optional Features	Ensure that you have Communication Manager version 2.0 (V12) or later.	G3 Version
	Ensure that the MLPP feature is on.	Multiple Level Precedence & Preemption

Screen name	Purpose	Fields
Feature Access Code (FAC)	Set up FACs for users to activate the MLPP features.	- Precedence Calling Access Code - All the fields in the WNDP Precedence Access Codes area
	Set up a FAC for users to answer a Precedence Call Waiting call from a single-line analog telephone.	CAS Remote Hold/Answer Hold-Unhold Access Code
Multiple Level Precedence & Preemption Parameters	Set up the system parameters for MLPP.	All
Class of Restriction	- Apply administration settings to all objects that share the same COR number. - Restrict the types of calls that a user can make and receive.	- Maximum Precedence Level - Preemptable - MLPP Service Domain - Lineload Control
Trunk Features	Assign a trunk group as a DSN termination telephone.	DSN Term
	[For Tie trunks – DS1 and analog only] Indicate if the precedence level is received or sent as digits (rotary pulses) or as DTMF signals (touchtone).	- Precedence Incoming - Precedence Outgoing
Station	Activate or deactivate Precedence Call Waiting for a telephone.	Precedence Call Waiting
	Assign a telephone as a hot line telephone for a precedence call or a WNDP call.	All the fields in the Hot Line Destination area
Abbreviated Dialing List	Set up the dialing list and specific dialed string to be used with a hot line telephone.	All
Console Parameters	Assign attendant queue priorities.	All the fields in the Queue Priorities area
Announcements/Audio Sources	Assign extensions for Precedence Calling announcements.	All
Precedence Routing Digit Analysis Table	Administer how the system analyzes Precedence Routing digits.	All

Screen name	Purpose	Fields
<i>Pattern Number</i>	Administer how the system handles route patterns for outgoing Precedence calls.	All
<i>Precedence Routing Digit Conversion Table</i>	Administer how the system takes digits of incoming calls, and converts the digits to local telephone numbers.	All

Precedence Calling

Assigning an MLPP feature access code

To assign a feature access code:

- 1 Type **change feature-access-codes**. Press **Enter**.
The system displays the *Feature Access Code (FAC)* screen.
- 2 Click **Next** until you see the MLPP Features area ([Figure 211](#)).

Figure 211: Feature Access Code (FAC) screen

change feature-access-codes	Page 6 of 6
FEATURE ACCESS CODE (FAC) MLPP Features	
Precedence Calling Access Code: 8	
W NDP PRECEDENCE ACCESS CODES:	
Flash Override Access Code: 90	
Flash Access Code: 91	
Immediate Access Code: 92	
Priority Access Code: 93	
Routine Access Code: 94	

- 3 In the Precedence Calling Access Code field, type a FAC that conforms to your dial plan.
- 4 Press **Enter** to save your changes.
- 5 Ensure that you notify all users of the assigned FAC.

Assigning Precedence Calling system parameters

To assign the Precedence Calling system parameters:

- 1 Type **change system-parameters mlpp**. Press **Enter**.
The system displays the *Multiple Level Precedence & Preemption Parameters* screen ([Figure 212, Multiple Level Precedence & Preemption Parameters screen](#), on page 835).

Figure 212: Multiple Level Precedence & Preemption Parameters screen

change system-parameters mlpp Page 1 of 1

MULTIPLE LEVEL PRECEDENCE & PREEMPTION PARAMETERS

ANNOUNCEMENTS

Blocked Precedence Level: 6801 Service Interruption: 6803

Unauthorized Precedence Level: 6802 Busy, Not Equipped: 6804

Vacant Code: 6805

PRECEDENCE CALLING-DIALED DIGIT ASSIGNMENT

Flash Override: 0 Flash: 1 Immediate: 2 Priority: 3 Routine: 4

Attendant Diversion Timing (sec): 60

Remote Attendant Route String:

Worldwide Numbering Dial Plan Active? y Default Route Digit: 0

Precedence Call Timeout (sec): 30

Line Load Control Restriction Level: 0

WNDP Emergency 911 Route String:

Default Service Domain: 1

ISDN Precedence Call Timeout (sec): 30

- 2 In the Precedence Calling-Dialed Digit Assignment fields, make any necessary changes.



CAUTION:

We highly recommend that you do not change the default Precedence Calling dialed digits unless you are coordinating this change with other companion networks in your system. If the Precedence Calling digits do not match across networks, the system does not properly process the calls. Each of the Precedence Calling digits must be different. You cannot use the same digit for two different precedence levels.

- Flash Override — **0–9** or blank (default is **0**)
- Flash — **0–9** or blank (default is **1**)
- Immediate — **0–9** or blank (default is **2**)
- Priority — **0–9** or blank (default is **3**)
- Routine — **0–9** or blank (default is **4**)
- Attendant Diversion Timing — **10 to 99** seconds or blank (default is blank)
- Remote Attendant Route String — 1 to 24 numeric digits or blank (default is blank). When you administer this string, use address digits only not FACs. For more information, see [Precedence Calling diversion scenarios](#) on page 817.
- Precedence Call Timeout — **10 to 60** seconds (default is **30**)
- Default Service Domain — **0 to 16777215**. This number defines the system service domain, and must be unique within a switching network. The system uses the system service domain to determine eligibility for precedence calling when interswitch precedence calls over non-ISDN trunks occur.

- 3 Press **Enter** to save your changes.

Assigning a maximum precedence level

Maximum precedence levels are assigned to Classes of Restriction (COR).

To assign a maximum precedence level to a COR:

- 1 Type **change cor *n***, where *n* is the number of a specific COR. Press **Enter**.
The system displays the **Class of Restriction** screen.
- 2 Click **Next** until you see the Maximum Precedence Level and MLPP Service Domain fields ([Figure 213, Class of Restriction screen](#), on page 836).

Figure 213: Class of Restriction screen

change cor 1	CLASS OF RESTRICTION	Page 2 of 4
	MF Incoming Call Trace? n	
	Brazil Collect Call Blocking? n	
	Block Transfer Display? n	
Block Enhanced Conference/Transfer Displays? y	Remote Logout of Agent? y	
	Station Lock COR: 1	
Outgoing Disconnect Trunk Timer (Minutes):		
Line Load Control: 3		
Maximum Precedence Level: ro	Preemptable? y	
MLPP Service Domain: 1		

- 3 In the Maximum Precedence Level field, enter one of the following values:
 - **fo** – Flash Override
 - **fl** – Flash
 - **im** – Immediate
 - **pr** – Priority
 - **ro** – Routine (default)
- 4 In the MLPP Service Domain field, enter a number from **0** to **16777215**.
This number defines the service domain for users and trunks to which this particular COR is assigned. The system uses the service domain to create a group of MLPP users or facilities, within which precedence calls can be made.
- 5 Press **Enter** to save your changes.

Assigning trunks

To assign trunks:

- 1 Type **add trunk-group *n***, where *n* is the number of a trunk group. Press **Enter**.
The system displays the **Trunk Features** screen.
- 2 Click **Next** until you see the DSN Term field ([Figure 214, Trunk Features screen](#), on page 837).

Figure 214: Trunk Features screen

add trunk-group 10		Page 2 of X	
TRUNK FEATURES			
ACA Assignment? n	Measured: none		
	Internal Alert? n	Maintenance Tests? y	
	Data Restriction? n		
	Glare Handling: none		
Used for DCS? n			
Suppress # Outpulsing? n			
Seize When Maintenance Busy: neither-end			
Incoming Tone (DTMF) ANI: no		Per Call CPN Blocking Code:	
Connected to CO? n		Per Call CPN Unblocking Code:	
US NI Delayed Calling Name Update? y			
DSN Term? y	Precedence Incoming? dtmf(a-d)	Precedence Outgoing? dtmf(a-d)	

- 3** Use the DSN Term field to identify the trunk group as a DSN termination telephone (the default is **n**).

If	Then
- You enter y in the DSN Term field, AND - the value in the Group Type field on page 1 of this screen is tie,	the system displays the Precedence Incoming and Precedence Outgoing fields.

These two fields define whether the precedence level for DTMF or Tone trunks is received or sent as digits (rotary pulses) or as DTMF signals (touch-tone).

If	Then
- You enter n in the DSN Term field, OR - You enter y in the DSN Term field, and the value in the Group Type field on page 1 of this screen is isdn, OR - You enter y in the DSN Term field, and the value in the Group Type field on page 1 of this screen is tie, and the value in both the Outgoing Dial Type and Incoming Dial Type fields on page 1 of this screen is mf2/6,	the system does not display the Precedence Incoming and Precedence Outgoing fields.

- 4** Press **Enter** to save your changes.

Assigning a hot line number

A hot line is a telephone extension that is automatically dialed when you pick up the receiver. On a single-line telephone, assign a hot line destination. Use a system, group, or personal list that is administered to the Hot Line Destination.

The Hot Line number must include:

- The Precedence Calling FAC
- The precedence level digit (see [Precedence Calling](#) on page 816)
- The destination telephone number

For example, when the system dials **807208451111**, the system processes the call as a Flash Override precedence voice call to extension **720-845-1111**.

- **8** is the Precedence Calling access code (FAC)
- **0** is the digit for Flash Override
- **720-845-1111** is the destination telephone number

The following example explains how to administer number 807208451111 as a hot line number.

To administer a hot line number:

- 1 Type **add station next**. Press **Enter**.
The system displays the **Station** screen.
- 2 In the Type field, enter an analog telephone type (like a **2500** telephone).
- 3 Enter information in the other fields to add this new telephone extension.
- 4 Click **Next** until you see the Hot Line Destination areas ([Figure 215, Station screen](#), on page 838).

Figure 215: Station screen

add station 807208451111

Page 3 of 3

STATION

SITE DATA

Room:

Jack:

Cable:

Floor:

Building:

Headset? n

Speaker? n

Mounting: d

Cord Length: 0

Set Color:

ABBREVIATED DIALING

List1: 1

List2:

List3:

HOT LINE DESTINATION

Abbreviated Dialing List Number (From above 1, 2, or 3):

Dial Code:

Line Appearance: call-appr

- 5 In the Abbreviated Dialing List Number (From above 1, 2, or 3) field, enter the Abbreviated Dialing list number that contains your Hot Line Destination number.
You set up the Hot Line Destination number – in this example, 807208451111 – in the *Abbreviated Dialing List* screen ([Figure 216](#)).

Figure 216: Abbreviated Dialing List screen

add abbreviated-dialing group next

Page 1 of 1

Abbreviated Dialing List

Group List: 1

Size (multiple of 5): 5

Group Name: _____

Program Ext: _____ Privileged? n

DIAL CODE

01: 807208451111

02: _____

03: _____

04: _____

05: _____

For more information on the *Abbreviated Dialing List* screen, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

- 6 In the Dial Code field, type the dial code that is associated with the Hot Line Destination number.
- 7 Press **Enter** to save your changes.

Assigning attendant queue priorities

To assign attendant queue priorities:

- 1 Type **change console-parameters**. Press **Enter**.
The system displays the *Console Parameters* screen.
- 2 Click **Next** until you see the MLPP Precedence Call area ([Figure 217, Console Parameters screen](#), on page 840).

Figure 217: Console Parameters screen

change console-parameters	Page 4 of 5 SPE A
CONSOLE PARAMETERS	
QUEUE PRIORITIES	
MLPP Precedence Call	
Flash Override: 2	
Flash: 3	
Immediate: 4	
Priority: 5	

- 3 Use the MLPP Precedence Call area to assign attendant queue priorities for precedence calls and non-precedence calls. This process determines how calls are queued to the attendant console. Depending on the priority you want for processing calls, you can change the system defaults. The defaults are:

- Flash Override: 2
- Flash: 3
- Immediate: 4
- Priority: 5



CAUTION:

By default, emergency access calls receive higher priority processing than MLPP Precedence Calls. You can change the order of priority, but be careful when designating emergency calls to equal or lower priority. Call types with equal priority enter the queue on a first-in, first-out basis.

Routine precedence calls are treated as normal calls and use the same queue priorities as non-emergency and non-MLPP calls.

- 4 Press **Enter** to save your changes.

Announcements for Precedence Calling

This feature uses the standard recorded announcements feature in Communication Manager. For additional information about administering recorded announcements, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Adding extensions

To assign extensions for Precedence Calling announcements:

- 1 Type **change announcements**. Press **Enter**.

The system displays the **Announcements/Audio Sources** screen ([Figure 218, Announcements/Audio Sources screen](#), on page 841).

Figure 218: Announcements/Audio Sources screen

change announcements										Page	1 of	16
ANNOUNCEMENTS/AUDIO SOURCES												
Ext.	Type	COR	TN	Name	Q	QLen	Pr	Rt	Port			
1: 6801	integrated	1	1	Call blocked announcement	n	N/A	n	16	01A14			
2: 6802	integrated	1	1	Unauthorized level annnc	n	N/A	n	16	01A14			
3: 6803	integrated	1	1	Service interrupt annnc	n	N/A	n	16	01A14			
4: 6804	integrated	1	1	Busy, not equipped annnc	n	N/A	n	16	01A14			
5: 6805	integrated	1	1	Vacant code announcement	n	N/A	n	16	01A14			
6:		1	1		n							
7:		1	1		n							
8:		1	1		n							
9:		1	1		n							
10:		1	1		n							
11:		1	1		n							
12:		1	1		n							
13:		1	1		n							
14:		1	1		n							
15:		1	1		n							
16:		1	1		n							

- 2 Assign extensions for the different MLPP recorded announcements. The extensions you use for recorded announcements must already be administered in your dial plan, but cannot be used for any other purpose (such as individual telephones or directory numbers).

- 3 Press **Enter** to save your changes.

In the next step, you use the announcement that you set up in this step.

Assigning announcement types

After you add the announcement extensions, you must designate what extensions are used for each of the announcement types. This administration is unique to the Announcements for Precedence Calling feature.

To assign attendant queue priorities:

- 1 Type **change system-paramters mlpp**. Press **Enter**.
The system displays the *Multiple Level Precedence & Preemption Parameters* screen ([Figure 212, Multiple Level Precedence & Preemption Parameters screen](#), on page 835).
- 2 Match the extensions that you set on the *Announcements/Audio Sources* screen (see [Adding extensions](#) on page 840) with the five announcements.
- 3 Click **Enter** to save your changes.

Recording announcements

Once you have designated the extensions that are used for recorded announcements, use the following procedures to record and test the announcements. You must record the announcements from the attendant console or from a telephone that has console permissions.

To record each announcement:

- 1** Go off-hook and dial the Announcement feature access code (FAC).
- 2** Dial the extension of the announcement you want to record.
If an announcement session is already in progress, or if a save or restore command is in progress, you hear reorder tone. Try again later.
- 3** Press **1** and record the announcement after the tone.
If the announcement already exists and is marked "protected" in the announcements screen, you hear intercept tone.

The following wording is suggested for each of the five announcements:

Blocked precedence call: "Equal or higher precedence calls have prevented completion of your call. Please hang up and try again later."

Unauthorized precedence level attempted: "The precedence level requested is not authorized for your line. Please use an authorized precedence level, or ask your operator for assistance."

Service interruption prevented call completion: "A service interruption has prevented the completion of your call. Please wait 30 seconds and try again. In case of emergency, call your operator."

Busy, not equipped for Preemption or Precedence Call Waiting: "The number you have dialed is busy and not equipped for Preemption or Precedence Call Waiting."

Vacant code announcement: "Your call cannot be completed as dialed. Please consult your directory and call again or ask your operator for assistance."

- 4** Hang up when you are finished recording the message.

NOTE:

The system records the sound of the receiver returning to the telephone cradle. So hang up gently, press **Drop**, or press the switchhook with your finger.

- 5** After waiting 15 seconds, dial the extension of the announcement you just recorded. Listen to the recording.
 - If you need to record the message again, repeat this procedure.
 - If the message is satisfactory, hang up and repeat this procedure to record the other announcements.

Deleting announcements

To delete a recorded announcement:

- 1** Go off-hook at a telephone and dial the Announcement feature access code (FAC).
- 2** Dial the extension of the announcement you want to delete.
- 3** Press **3**.
The announcement is deleted.
- 4** Hang up.
- 5** Type **change announcements** to delete the announcement extension (see [Adding extensions](#) on page 840).

Saving announcements

If your system uses the TN750 or TN750B circuit packs, you must manually save the announcements recorded on those circuit packs. If you do not save the announcements, all announcements that were recorded since the last save are lost if:

- the system loses power, or
- the TN750 or TN750B circuit packs are removed from the system

For procedures to save announcements, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

If your system uses only the TN750C circuit pack, saving the announcements is not required. The TN750C has on-board memory for all announcements.

The announcements on a TN2501AP circuit pack and virtual VAL announcements on the S8300 are not saved using system administration. You can back up the announcement to a PC. For procedures to back up the announcements, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Precedence Call Waiting

Enabling Precedence Call Waiting for a telephone

To enabling Precedence Call Waiting for a telephone:

- 1 Type **change station *n***, where *n* is the telephone extension you want to enable. Press **Enter**.
The system displays the **Station** screen.
- 2 Click **Next** until you see the Precedence Call Waiting field.
- 3 Enable or disable Precedence Call Waiting for each telephone. The default assignment for each telephone is **y** (enabled).
- 4 Press **Enter** to save your changes.

Setting the Precedence Call timeout

To set the Precedence Call Waiting timeout:

- 1 Type **change system-paramters mlpp**. Press **Enter**.
The system displays the **Multiple Level Precedence & Preemption Parameters** screen ([Figure 212, Multiple Level Precedence & Preemption Parameters screen](#), on page 835).
- 2 In the Precedence Call Timeout (sec) field, set the number of seconds before a precedence call is timed out.
The valid values are **10 to 60** seconds, with a default of **30** seconds.
- 3 In the ISDN Precedence Call Timeout (sec) field, set the number of seconds before an ISDN precedence call is timed out.
The valid values are **10 to 60** seconds, with a default of **30** seconds.
- 4 Press **Enter** to save your changes.

Assigning feature access codes

For users on single-line analog telephones, you must assign a feature access code to answer a Precedence Call Waiting call.

To assign a feature access code:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the *Feature Access Code (FAC)* screen ([Figure 219, Feature Access Code \(FAC\) screen](#), on page 844).

Figure 219: Feature Access Code (FAC) screen

change feature-access-codes	Page 1 of 8
FEATURE ACCESS CODE (FAC)	
Abbreviated Dialing List1 Access Code: #01	
Abbreviated Dialing List2 Access Code: #02	
Abbreviated Dialing List3 Access Code: #03	
Abbreviated Dial - Prgm Group List Access Code: #04	
Announcement Access Code: #05	
Answer Back Access Code: #14	
Attendant Access Code:	
Auto Alternate Routing (AAR) Access Code: 8	
Auto Route Selection (ARS) - Access Code 1: *9	Access Code 2: *33
Automatic Callback Activation: #55	Deactivation: *55
Call Forwarding Activation: #44	Deactivation: *44
Call Park Access Code: #76	
Call Pickup Access Code: #33	
CAS Remote Hold/Answer Hold-Unhold Access Code: #57	
CDR Account Code Access Code:	
Change COR Access Code:	
Change Coverage Access Code:	
Contact Closure Open Code:	Close Code:
Contact Closure Pulse Code:	

- 2 In the CAS Remote Hold/Answer Hold-Unhold Access Code field, assign a FAC that matches your dial plan.
- 3 Press **Enter** to save your changes.
- 4 Ensure that you notify all users of the assigned FAC.

Precedence Routing

This section contains procedures for administering Precedence Routing when the local and remote DSN nodes are both Avaya communication servers. Call your Avaya representative for help with:

- trunk administration when connecting local DS1 trunks to a DSN node that is using an Avaya communication server
- administration procedures when connecting local DS1 trunks to a DSN node that uses a Nortel switch
- administration procedures when connecting local DS1 trunks to a DSN node that uses a Siemens switch

Assigning digit analysis

Digit analysis determines what routes are used for outgoing calls based on the digits dialed.

To assign digit analysis:

- 1 Type **change precedence-routing analysis *n***, where *n* is the digit or digits being analyzed. Press **Enter**.

The system displays the **Precedence Routing Digit Analysis Table** screen ([Figure 220, Precedence Routing Digit Analysis Table screen](#), on page 845).

Figure 220: Precedence Routing Digit Analysis Table screen

change precedence-routing analysis 10				Page	1 of 2	SPE A
PRECEDENCE ROUTING DIGIT ANALYSIS TABLE				Percent Full: 22		
Dialed String	Total Min	Total Max	Route Pattern	Preempt Method		
002383	9	9	36	group		
002385	9	9	35	group		
002388	9	9	86	group		
003032383	12	12	36	group		
003032388	12	12	86	group		
003033383	12	12	34	group		
003033388	12	12	84	group		
003034383	12	12	32	group		
003034388	12	12	82	group		
003035383	12	12	30	group		
003035388	12	12	80	group		
003383	9	9	34	group		
003385	9	9	33	group		
003388	9	9	84	group		
004383	9	9	32	group		

Except for the **Preempt Method** field, the digit analysis administration is the same as ARS/AAR digit analysis.

- 2 Format the **Dialed String** field for routing DSN numbers:

- For non-WNDP dialing, enter the precedence digit – usually 0-4 – and the address digits.
- For WNDP dialing, enter the precedence digit – usually 0-4 – the route code, and the address digits.

An **x** in the digit string is a wildcard that matches any single digit.

The **Preempt Method** field has two possible values: **group** and **route**. The default preemption is **group**.

Group preemption

With group preemption:

- a The system checks the first trunk group in the route pattern to determine if any trunks are idle. If an idle trunk is found, the call is connected.
- b If no trunks are idle, the system checks the same trunk group to determine if any trunks are preemptable. If a preemptable trunk is found, the current call is preempted and the new call is connected.

- c** If no trunks are idle or preemptable, the system checks the next trunk group to determine if any trunks are idle. If an idle trunk is found, the call is connected.
- d** If no trunks are idle, the system checks the same trunk group to determine if any trunks are preemptable. If a preemptable trunk is found, the current call is preempted and the new call is connected.
- e** If an idle trunk or a preemptable trunk is not found within the Precedence Call Timeout interval, the caller hears the Blocked Precedence recorded announcement or reorder tone (see [Announcements for Precedence Calling](#) on page 824). Calls with Routine precedence cannot preempt any other calls. If a call with Routine precedence fails to find an idle trunk, the caller receives busy tone.

For example, trunk groups 1, 2, and 3 are set up as follows:

- Trunk group 1 has two trunk members active with Flash and Flash Override precedence calls.
- Trunk group 2 has two trunk members active with Immediate and Priority precedence calls.
- Trunk group 3 has two trunk members, both idle.

A new call is made with the Flash precedence level. The call is processed as follows:

- a** Trunk group 1 is checked for an idle trunk and one is not found. Trunk group 1 is then checked for a preemptable active call and one is not found.
- b** Trunk group 2 is checked for an idle trunk and one is not found. Trunk group 2 is then checked for a preemptable active call. Both calls are preemptable. The new Flash call preempts the first trunk member that the system finds with a lower precedence level in trunk group 2. In this example, the new Flash call preempts the Immediate call.
- c** Trunk group 3 is never checked even though it has idle trunks.

Route preemption

With route preemption:

- a** The system checks each trunk group in the route pattern to determine if any trunks are idle. If an idle trunk is found, the call is connected.
- b** If no trunks are idle, the system checks each trunk group in the pattern to determine if any trunks are preemptable. If a preemptable trunk is found, the current call is preempted and the new call is connected.
- c** If an idle trunk or a preemptable trunk is not found within the Precedence Call Timeout interval, the caller hears the Blocked Precedence recorded announcement or reorder tone (see [Announcements for Precedence Calling](#) on page 824). Calls with Routine precedence cannot preempt any other calls. If a call with Routine precedence fails to find an idle trunk, the caller receives busy tone.

For example, trunk groups 1, 2, and 3 are set up as follows:

- Trunk group 1 has two trunk members active with Flash and Flash Override precedence calls.
- Trunk group 2 has two trunk members active with Immediate and Priority precedence calls.
- Trunk group 3 has two trunk members, both idle.

A new call is made with the Flash precedence level. The call is processed as follows:

- a** Trunk group 1 is checked for an idle trunk and one is not found.

- b** Trunk group 2 is checked for an idle trunk and one is not found.
 - c** Trunk group 3 is checked for an idle trunk and one is found. The new Flash call is completed using the first idle trunk.
- 3** Press **Enter** to save your changes.

Assigning route patterns

The system uses these routing patterns for outgoing calls.

To assign route patterns:

- 1** Type **change route-pattern *n***, where *n* is a route pattern from the **Precedence Routing Digit Analysis Table** screen. Press **Enter**.
- The system displays the **Pattern Number** screen ([Figure 221, Pattern Number screen](#), on page 847).

Figure 221: Pattern Number screen

change route-pattern 36										Page 1 of 1	
Pattern Number: 36											
Grp. No.	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del	Inserted Dgts	DCS/ QSIG Intw	IXC		
1: 15		0					1	n	user		
2: 12		1					1	n	user		
3: 7		0					1	n	user		
4: 22		0					1	n	user		
5:								n	user		
6:								n	user		
BCC VALUE		TSC		CA-TSC		ITC		BCIE		Service/Feature BAND	
0	1	2	3	4	W	Request		No. Dgts	Numbering Format		LAR
										Subaddress	
1:	y	y	y	y	y	n	n	rest		none	
2:	y	y	y	y	y	n	n	rest		none	
3:	y	y	y	y	y	n	n	rest		none	
4:	y	y	y	y	y	n	n	rest		none	
5:	y	y	y	y	y	n	n	rest		none	
6:	y	y	y	y	y	n	n	rest		none	

- 2** For DSN trunks that have the **Precedence Mode Outgoing** field set for DTMF, you must delete one digit. Otherwise, the system sends the precedence level digit twice.
- 3** Press **Enter** to save your changes.

Assigning digit conversion

Digit conversion takes digits that were dialed on incoming calls, and converts the digits to local telephone numbers, usually extensions.

To assign the Precedence Routing digit conversion:

- 1 Type **change precedence-routing digit-conversion**. Press **Enter**.

The system displays the **Precedence Routing Digit Conversion Table** screen ([Figure 222, Precedence Routing Digit Conversion Table screen](#), on page 848).

Figure 222: Precedence Routing Digit Conversion Table screen

change precedence-routing digit-conversion 0					Page 1 of 2	
PRECEDENCE ROUTING DIGIT CONVERSION TABLE					Percent Full: 11	
Matching Pattern	Min	Max	Del	Replacement String	Net	Conv
x2386	8	8	4		ext	n
x3032386	11	11	7		ext	n
x3033386	11	11	7		ext	n
x3034386	11	11	7		ext	n
x3035386	11	11	7		ext	n
x3386	8	8	4		ext	n
x4386	8	8	4		ext	n
x5386	8	8	4		ext	n
x6	5	5	1		ext	n
xx2386	9	9	5		ext	n
xx3032386	12	12	8		ext	n
xx3033386	12	12	8		ext	n
xx3034386	12	12	8		ext	n
xx3035386	12	12	8		ext	n

- 2 Format the Matching Pattern field for routing DSN numbers:

- For non-WNDP dialing, enter the precedence digit – usually 0-4 – and the address digits.
- For WNDP dialing, enter the precedence digit – usually 0-4 – the route code, and the address digits.

An **x** in the digit string is a wildcard that matches any single digit.

- 3 In the Net fields, type **ext** or **pre**.

- **ext** stands for extension, and uses ARS of AAR tables for routing the call.
- **pre** stands for precedence routing, and uses the Precedence Analysis Tables for routing the call.

- 4 Press **Enter** to save your changes.

Dual Homing

The administration of Dual Homing is done when you administer the Precedence Routing feature (see [Precedence Routing](#) on page 844).

End Office Access Line Hunting

The administration of End Office Access Line Hunting is done when you administer the Precedence Routing feature (see [Precedence Routing](#) on page 844).

Preemption

Assigning Preemption to a COR

To assign preemption to a COR:

- 1 Type **change cor *n***, where *n* is the number of a specific COR. Press **Enter**.
The system displays the *Class of Restriction* screen.
- 2 Click **Next** until you see the Preemptable field.
- 3 In the Preemptable field, define whether extensions or trunks assigned to this COR can be preempted from their current calls (the default is set to **y**).
- 4 Press **Enter** to save your changes.

Assigning trunks for Preemption

For the Preemption feature, enabling a trunk as a DSN termination telephone guarantees that the trunk can accept the Preemption signaling over the DSN. To assign trunks, see [Assigning trunks](#) on page 836.

Setting the Precedence Call timeout

See [Assigning Precedence Calling system parameters](#) on page 834.

Line Load Control

Line Load Control (LLC) is assigned on a system-wide basis and on a COR basis.

Assigning the LLC level for the system

- 1 Type **change system-parameters mlpp**. Press **Enter**.
The system displays the *Multiple Level Precedence & Preemption Parameters* screen ([Figure 212, Multiple Level Precedence & Preemption Parameters screen](#), on page 835).
- 2 Set the LLC level for the system. The options are as follows:
 - **0** – Feature not active (no restrictions – default)
 - **2** – Restrict stations with a COR assigned to LLC level 2, 3, or 4
 - **3** – Restrict stations with a COR assigned to LLC level 3 or 4
 - **4** – Restrict stations with a COR assigned to LLC level 4
- 3 Press **Enter** to save your changes.

Assigning the LLC to a COR

- 1 Type **change cor *n***, where *n* is the number of a specific COR. Press **Enter**.
The system displays the *Class of Restriction* screen.
- 2 Click **Next** until you see the Line Load Control field ([Figure 213, Class of Restriction screen](#), on page 836).

- 3 Set the Line Load Control level for each COR. The options are as follows:
 - 1 – LLC Level 1 (cannot be restricted by LLC – default)
 - 2 – LLC Level 2
 - 3 – LLC Level 3
 - 4 – LLC Level 4
- 4 Press **Enter** to save your changes.

Worldwide Numbering and Dialing Plan

Assigning WNDP system parameters

- 1 Type **change system-parameters mlpp**. Press **Enter**.

The system displays the *Multiple Level Precedence & Preemption Parameters* screen ([Figure 212, Multiple Level Precedence & Preemption Parameters screen](#), on page 835).
- 2 Administer the following fields:
 - Worldwide Numbering Dial Plan Active – **y** or **n** (default is **n**)
 - Default Route Digit – This field displays when the Worldwide Numbering Dial Plan Active field is enabled. You must enter a valid digit in this field. Valid entries are:
 - 0 = Voice call (the default value)
 - 1 = Circuit switched data call
 - 2 = Satellite avoidance call
 - 3 = (reserved)
 - 4 = (reserved)
 - 5 = Hotline voice grade call
 - 6 = Hotline data grade call
 - 7 = (reserved)
 - 8 = (reserved)
 - 9 = (reserved)
 - WNDP Emergency 911 Route String – 1 to 24 numeric digits or blank (default is blank). See [Interactions for Multiple Level Precedence and Preemption](#) on page 854 for more information. Valid entries for this field can be a trunk access code, the AAR or ARS access code, a WNDP access code, or an extension (for example, the firehouse at a base that handles emergency calls). If you use a WNDP access code, use the access code for the lowest precedence calling level in the system.
- 3 Press **Enter** to save your changes.

Assigning WNDP feature access codes

Administer the WNDP Precedence FACs only when using WNDP dialing and not precedence dialing.

To assign a feature access code:

- 1 Type **change feature-access-codes**. Press **Enter**.
The system displays the *Feature Access Code (FAC)* screen.
- 2 Click **Next** until you see the WNDP Precedence Access Codes area ([Figure 211, Feature Access Code \(FAC\) screen](#), on page 834).

NOTE:

A value in the Precedence Calling Access Code field must also be defined even when using WNDP dialing. The Precedence Calling FAC is used when administering Precedence Routing trunks. For more information, see [Assigning an MLPP feature access code](#) on page 834.

- 3 In the WNDP Precedence Access Codes area, type a two-digit FAC, each beginning with the number **9**, in each field that conforms to your dial plan.
- 4 Press **Enter** to save your changes.
- 5 Ensure that you notify all users of the assigned FACs.

Assigning a Hot Line number

A hot line is a telephone extension that is automatically dialed when you pick up the receiver. On a single-line telephone, assign a hot line destination using a system, group, or personal list that is administered to the Hot Line Destination.

The Hot Line number must include:

- The WNDP Dialing two-digit feature access code for the precedence level that you want.
- As an option, the number **1** as the Route Code setup digit. See [Worldwide Numbering and Dialing Plan](#) on page 828. If you do not use a Route Code setup digit, the system uses the default digit zero (**0**).
- If you use a Route Code setup digit, the next number is the routing digit. For example, **5** is a Hot Line voice call, and **6** is a Hot Line data call. See [Worldwide Numbering and Dialing Plan](#) on page 850.
- The destination telephone number.

For example, when the system dials **90157208451111**, the system processes the call as a Flash Override WNDP voice call to extension **720-845-1111**.

- **90** is the two-digit WNDP Dialing feature access code (FAC) indicating Flash Override
- **1** is the optional Route Code setup digit
- **5** is the routing digit for a voice hot line call
- **720-845-1111** is the destination telephone number

The following example explains how to administer number 90157208451111 as a hot line number.

To administer a hot line number:

- 1 Type **add station next**. Press **Enter**.
The system displays the **Station** screen.
- 2 In the Type field, enter an analog telephone type (like a **2500** telephone).
- 3 Enter information in the other fields to add this new telephone extension.
- 4 Click **Next** until you see the Hot Line Destination area ([Figure 215, Station screen](#), on page 838).
- 5 In the Abbreviated Dialing List Number (From above 1, 2, or 3) field, enter the Abbreviated Dialing list number that contains your Hot Line Destination number.
You set up the Hot Line Destination number – in this example, 90157208451111 – in the **Abbreviated Dialing List** screen. For more information on the **Abbreviated Dialing List** screen, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.
- 6 In the Dial Code field, type the dial code that is associated with the Hot Line Destination number.
- 7 Press **Enter** to save your changes.

Assigning attendant queue priorities

See [Assigning attendant queue priorities](#) on page 839.

Reports for Multiple Level Precedence and Preemption

The following reports provide information about the Multiple Level Precedence and Preemption feature:

- None

Considerations for Multiple Level Precedence and Preemption

This section provides information about how the Multiple Level Precedence and Preemption feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Multiple Level Precedence and Preemption under all conditions. The following considerations apply to Multiple Level Precedence and Preemption:

Precedence Calling

Except for a G350 or G700 Media Gateway supported by an S8300, S8500, or S8700 Media Server, a tone clock circuit pack is required for this feature. On the DEFINITY server, all tone clocks must be TN2182B or later, except when using MF 2/6 trunks, which require the TN2182C, Vintage 2 or later.

Announcements for Precedence Calling

When using the announcement capabilities on a G650, SCC1, or MCC1 Media Gateway, use the TN2501AP or TN750 circuit packs for recording announcements on a:

- DEFINITY server
- S8100 Media Server
- S8500 Media Server
- S8700 Media Server

You can also use existing analog announcement equipment with the MLPP features.

If you have multiple integrated announcement circuit packs, only one of those can be a TN750 or TN750B. Any additional circuit packs must be the TN2501AP or TN750C.

Precedence Call Waiting

Except for an S8300 Media Server, a Tone Clock circuit pack is required for this feature. On the DEFINITY server and IP600 IP Server, all tone clocks must be TN2182B or later, except when using MF 2/6 trunks, which require the TN2182C, Vintage 2 or later.

Precedence Call Waiting can be assigned to all models of telephones, including IP hardphones and softphones.

Precedence Routing

The routing administered with Precedence Routing uses the same capacity tables as the ARS feature (patterns, analysis tables, and so on). You can view the real-time capacity usage with the **change precedence-routing analysis** command. The Percent Full field displays how much of the available capacity is being used for routing information.

H.323 IP trunks do not support data calls using any type of modem or data module. Do not administer H.323 IP trunks for users that make data calls over trunk facilities. H.323 IP trunks can be used for voice calls.

Dual Homing

- None

End Office Access Line Hunting

- None

Preemption

A tone clock circuit pack is required for this feature. All tone clocks must be TN2182B or later, except when using MF 2/6 trunks, which require the TN2182C, Vintage 2 or later.

The call progress tones used for Preemption are a fixed tone and pattern. The tones cannot be changed using the **change system-parameters country-options** command.

Line Load Control

When a system reload occurs, the LLC system-level settings revert to the default factory setting. The default factory setting is LLC level 0 (no restrictions). Normal telecommunications service is restored after a system reload.

The LLC COR settings, however, are saved in translations. The settings do not revert to the factory defaults if the settings were saved in translations.

Worldwide Numbering and Dialing Plan (WNDP)

The Route Control Digit of WNDP is not available as part of the MLPP feature set. This is not the same as the Route Code digit.

Any destination telephone number that starts with a 1 (such as the extension 1500) must be dialed as follows: 1, Route Code digit, destination number. For example, dial **1x1500**, where **x** is the Route Code digit.

H.323 IP trunks do not support data calls. Do not administer H.323 IP trunks for users that make data calls over trunk facilities. H.323 IP trunks can be used for voice calls.

Interactions for Multiple Level Precedence and Preemption

This section provides information about how the Multiple Level Precedence and Preemption feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Multiple Level Precedence and Preemption in any feature configuration.

Precedence Calling

- Attendant Vectoring

Attendant Vectoring uses the Call Vectoring feature to provide flexible routing of attendant-seeking calls. When the system accesses an attendant VDN, the call can be answered by the attendant, routed to an announcement, or routed to the voice mailbox of the assigned night station. This process reduces the chance of precedence calls remaining in the attendant queue, specifically

where the attendant has not answered all calls in queue and the console is placed in night mode. Attendant Vectoring can be purchased as a stand-alone feature by contacting your Avaya representative.

For more information about Attendant Vectoring, see *DEFINITY ECS Call Vectoring/EAS Guide*, 555-230-521.

- **Call Coverage**

Calls above Routine precedence do not follow administered coverage paths. The calls ring until the Timeout for Precedence Calls expires, and the call goes to a console or night station. If the called party is on an active call and Preemption is enabled, the call is preempted.

- **Call Detail Recording (CDR)**

No separate CDR field is supplied for the precedence level of a call. No separate CDR field creates an incompatibility between current call accounting software and the new call record format. Using the current call record format, the precedence level of a call can be determined by examining the call record for the Precedence Calling FAC. If the call is a precedence call, the first digit of the address dialed indicates the precedence level of the call. If WNDP is active, only the FAC needs to be examined as the precedence is implied from the FAC. The CDR feature does not record the precedence level for a station-to-station call.

CDR can be administered to record either the dialed digits or the outpulsed digits. In the case of Precedence Routing, the outpulsed digits can appear dramatically different from the dialed digits. The precedence level digit might not be recorded. Keep this in mind when viewing CDR records.

- **Conference**

When two calls are merged during a conference, the precedence level of a call is set to the highest active precedence level.

- **Hunting**

When administering a hunt group with preemption, set the Maximum Preemption Level and Preemptable fields in the COR form. The hunt group Group Type must use circular or ucd-mia queuing, and the ACD, Queue, and Vector fields must be set to n.

- **Night Service**

When the attendant console goes into night mode, precedence calls can be answered using a night station, a night or day/night console, or Trunk Answer Any Station (TAAS). If there are calls in queue when the attendant console goes into Night Service, those queued calls are diverted to a night or day/night console or TAAS based on the attendant queue priorities (see Assigning attendant queue priorities on page 41).

- **Preemption**

When a precedence call attempts to preempt an existing call, call progress tones or the "blocked precedence call" announcement indicates why the call did not complete. The following table shows how precedence calls are processed depending on the precedence level of the call and the precedence level of the preempted trunk.

Precedence level of the call	Precedence level of the DSN trunk call being preempted	Call treatment
Flash Override	Flash Override	Recorded announcement or busy tone
Flash	Flash Override	Recorded announcement or busy tone

Precedence level of the call	Precedence level of the DSN trunk call being preempted	Call treatment
Immediate	Flash Override	Recorded announcement or busy tone
Priority	Flash Override	Recorded announcement or busy tone
Routine	Flash Override	Busy tone
Flash Override	Flash	Call completes normally
Flash	Flash	Recorded announcement or busy tone
Immediate	Flash	Recorded announcement or busy tone
Priority	Flash	Recorded announcement or busy tone
Routine	Flash	Busy tone
Flash Override	Immediate	Call completes normally
Flash	Immediate	Call completes normally
Immediate	Immediate	Recorded announcement or busy tone
Priority	Immediate	Recorded announcement or busy tone
Routine	Immediate	Busy tone
Flash Override	Priority	Call completes normally
Flash	Priority	Call completes normally
Immediate	Priority	Call completes normally
Priority	Priority	Recorded announcement or busy tone
Routine	Priority	Busy tone
Flash Override	Routine	Call completes normally
Flash	Routine	Call completes normally
Immediate	Routine	Call completes normally
Priority	Routine	Call completes normally
Routine	Routine	Busy tone

- **Restrict Last Appearance**

If the Restrict Last Appearance telephone option is enabled and there is only one idle call appearance available on the telephone when a precedence call is made to that telephone, callers using Routine precedence hear busy tone. Callers using any other precedence level ring the restricted last call appearance.

If the Restrict Last Appearance telephone option is not enabled and there is only one idle call appearance available on the telephone when a precedence call is made to that telephone, callers using any precedence level ring that last call appearance.

- **Send All Calls**

Calls above Routine precedence do not follow administered coverage paths. The calls rings until the Timeout for Precedence Calls expires, and the call goes to a console or night station.

- **Transfer**
When two calls are merged during a transfer, the precedence level of a call is set to the highest active precedence level.
- **Worldwide Numbering and Dialing Plan (WNDP)**
When WNDP is enabled, users must dial a FAC for the precedence level they want to use. Users cannot use the Precedence Calling FAC. In addition, the route code function and implied precedence level are provided.

When WNDP is disabled, users must dial the Precedence Calling FAC, followed by the precedence level (0-4). The WNDP FACs can be administered, but cannot be used.

Announcements for Precedence Calling

- None

Precedence Call Waiting

- **Attendant Console**
Precedence Call Waiting calls from attendant consoles or telephones with console permissions are not allowed. Calls from an attendant console cannot camp onto a call with Precedence Call Waiting. The attendant console user hears a recorded announcement.
- **Automatic Callback**
If the Automatic Callback feature is activated and Precedence Call Waiting is attempted, the caller hears a recorded announcement.
- **Call Forwarding**
An extension can have Precedence Call Waiting and Call Forwarding active at the same time. If the user is active on a call and another call comes in, the called party hears Precedence Call Waiting tone. The call is forwarded after the timeout. Any other calls that arrive during the timeout period go immediately to the forwarded telephone.
- **Call Pickup**
If a member of a pickup group active on a call receives Precedence Call Waiting, other members of the pickup group cannot pick up the call.
- **Call Waiting**
For a Routine Precedence Call, a user on an active call hears the standard Call Waiting tone.

Precedence Call Waiting is denied if the called party already has one call currently waiting in queue (either Standard Call Waiting or Precedence Call Waiting).
- **Data Privacy**
Precedence Call Waiting cannot be applied to a line with Data Privacy.
- **Data Restriction**
Precedence Call Waiting cannot be applied to a line with Data Restriction.
- **Line Load Control**
If the LLC feature restricts a telephone, that user must hang up to answer a Precedence Call Waiting call.

- Preemption

Higher level precedence calls always preempt lower precedence calls regardless of how Precedence Call Waiting is administered.
- Tenant Service Partitioning

This feature operates as described, except that timeout redirection do not occur. The call continues to ring at the called extension.

Precedence Routing

- General

Precedence Routing allows calls with precedence higher than Routine to terminate to:

 - trunks
 - telephones
 - attendant consoles
 - hunt groups
 - recorded announcements

Precedence Routing calls cannot terminate at Vector Directory Numbers (VDNs) and Terminating Extension Groups (TEGs).
- Call Detail Recording (CDR)

No separate CDR field is supplied for the precedence level of a call. No separate CDR field creates an incompatibility between current call accounting software and the new call record format. Using the current call record format, the precedence level of a call can be determined by examining the call record for the Precedence Calling FAC. If the call is a precedence call, the first digit of the address dialed indicates the precedence level of the call. If WNDP is active, only the FAC needs to be examined as the precedence is implied from the FAC. The CDR feature does not record the precedence level for a station-to-station call.

CDR can be administered to record either the dialed digits or the outpulsed digits. In the case of Precedence Routing, the outpulsed digits can appear dramatically different from the dialed digits. With outpulsed digits, the precedence level digit might not be recorded based on the characteristics of the trunk group. With dialed digits, the precedence level digit is recorded. Keep this in mind when viewing CDR records.
- Shortcut Dialing

When using the Shortcut Dialing feature over DSN trunks, the incoming Shortcut Dialing digit analysis is administered using the Precedence Routing analysis tables instead of the ARS analysis tables. See [Assigning digit analysis](#) on page 845 for more information.
- Traveling Class Marks

Precedence Routing passes all Traveling Class Mark (TCM) information over DSN and non-DSN trunks.

Dual Homing

- None

End Office Access Line Hunting

- None

Preemption

- General

Calls can be preempted that terminate at:

- trunks
- telephones
- attendant consoles
- non-queued hunt groups
- Vector Directory Numbers (VDNs)

Since Precedence Routing does not allow precedence calls to terminate at queued hunt groups or Terminating Extension Groups (TEGs), other calls to these facilities cannot be preempted.

- Adjunct Switch Applications Interface (ASAI)

ASAI is notified if a call is preempted and disconnected from the current call.

- AUDIX system

The AUDIX system is notified if a call is preempted and disconnected from the current call.

- Call Detail Recording (CDR)

CDR has two records for a preempted call: the original call, and the new call after the preemption.

- Call Management System (CMS)

CMS is notified if a call is preempted and disconnected from the current call.

- Call Coverage

A call being redirected to a coverage point cannot be preempted.

- Call Pickup

A call using Call Pickup cannot be preempted.

- Code Calling Access

A call using Code Calling Access cannot be preempted.

- CONVERSANT[®] system

A CONVERSANT[®] system is notified if a call is preempted and disconnected from the current call.

- Group Paging

A call that is part of a group page cannot be preempted.

- Loudspeaker Paging

A call using Loudspeaker Paging cannot be preempted.

- Malicious Call Trace

A call using Malicious Call Trace cannot be preempted.

- Modem Pooling

A call using a modem pool cannot be preempted.

- Personal CO Line

A call using a personal CO line cannot be preempted.

- Precedence Calling

When a precedence call attempts to preempt an existing call, call progress tones or the "blocked precedence call" announcement indicates why the call did not complete. The following table shows how precedence calls are processed depending on the precedence level of the call and the precedence level of the preempted trunk:

Precedence level of the call	Precedence level of the DSN trunk call being preempted	Call treatment
Flash Override	Flash Override	Recorded announcement or busy tone
Flash	Flash Override	Recorded announcement or busy tone
Immediate	Flash Override	Recorded announcement or busy tone
Priority	Flash Override	Recorded announcement or busy tone
Routine	Flash Override	Busy tone
Flash Override	Flash	Call completes normally
Flash	Flash	Recorded announcement or busy tone
Immediate	Flash	Recorded announcement or busy tone
Priority	Flash	Recorded announcement or busy tone
Routine	Flash	Busy tone
Flash Override	Immediate	Call completes normally
Flash	Immediate	Call completes normally
Immediate	Immediate	Recorded announcement or busy tone
Priority	Immediate	Recorded announcement or busy tone
Routine	Immediate	Busy tone
Flash Override	Priority	Call completes normally
Flash	Priority	Call completes normally
Immediate	Priority	Call completes normally
Priority	Priority	Recorded announcement or busy tone
Routine	Priority	Busy tone
Flash Override	Routine	Call completes normally
Flash	Routine	Call completes normally
Immediate	Routine	Call completes normally
Priority	Routine	Call completes normally
Routine	Routine	Busy tone

- Precedence Call Waiting
Regardless how you administer Precedence Call Waiting, higher level precedence calls always preempt lower precedence calls.
- Radio Paging
You cannot preempt a call using Radio Paging.
- Recorded Announcements
You cannot preempt a call connected to a recorded announcement.
- Secondary Extension
You cannot preempt a call using a secondary extension.
- Transient calls
You cannot preempt calls in a transient mode (ringing, on hold).

Line Load Control

- General
Since the LLC feature restricts telephones from originating calls, features that require dial tone or a new call appearance for activation are not available when LLC is restricting the telephone. Some of those features include:
 - Call Forwarding
 - Call Pickup
 - Conference
 - TransferFeature activation using buttons where dial tone is not required is still allowed (such as Send All Calls, Inspect, or Integrated Directory).
- Bridged Call Appearance
The LLC feature restricts originating new calls from all call appearances on a telephone, including bridged appearances. A telephone that is restricted by the LLC feature, and that has a bridged appearance of an extension whose telephone is *not* restricted, can bridge onto an active call, but cannot originate a new call using that bridged extension.
A telephone that is *not* restricted by the LLC feature, and that has a bridged appearance of an extension whose telephone *is* restricted, can originate a new call using that bridged extension.
- Call Park
A user on a call becomes restricted by the LLC feature. The user can park the call, but cannot retrieve the call until the LLC restriction is removed. Another user that is not currently restricted by the LLC feature can retrieve the call.
- Call Waiting
A user restricted by the LLC feature must hang up to answer a Call Waiting call. The LLC feature does not restrict incoming calls.
- Hold
Telephones that are restricted by the LLC feature, and that are on an active call, can place a call on hold and later retrieve the call on hold.

- Precedence Call Waiting

A user whose telephone is restricted by the LLC feature must hang up to answer a Precedence Call Waiting call.

Worldwide Numbering and Dialing Plan (WNDP)

- Emergency 911 Calling

If WNDP dialing is administered to use the digits "91" as a FAC, this FAC conflicts with dialing 911 to reach an emergency service agency. If a user dials 911 thinking the call is connected to an emergency service agency, the call does not go through because the system is waiting for more digits after dialing 91.

To work around this interaction, you can take all or any the following actions:

- Instruct users to dial the ARS FAC followed by 911. For example, if the ARS FAC is 8, a user can dial **8911**. This option works for any emergency numbers (for example, 999 in the United Kingdom).
- Instruct users to dial their assigned WNDP FAC followed by 911. For example, if one of the WNDP FACs they use is 92, a user can dial **92911**. This option works for any emergency numbers (for example, 999 in the United Kingdom).
- Administer a WNDP Emergency 911 Route String (see [Assigning WNDP system parameters](#) on page 850 for an example). This route string is outpulsed when a user dials either 911 and waits for the interdigit timeout, or dials 911 followed by #. This dialing option only works when the WNDP Flash FAC is 91.

If the telephone that you use for an emergency call does not have adequate calling permissions, the emergency call does not go through. This situation can happen in the following conditions:

- The Facility Restriction Level (FRL) of the telephone is not high enough.
- The precedence calling level of the telephone is not high enough.
- The telephone is not allowed to use a higher precedence level for the call.
- There are no available trunk facilities and the precedence level of the call is not high enough to preempt another call.
- The hop limit is exceeded when call is routed over tandem trunks.

- Hunting

When administering a hunt group with preemption, set the **Maximum Preemption Level** and **Preemptable** fields in the COR form. The hunt group **Group Type** must use circular or ucd-mia queuing, and the **ACD**, **Queue**, and **Vector** fields must be set to **n**.

- Precedence Calling

When WNDP is enabled, users must dial a FAC for the precedence level they want to use. Users cannot use the Precedence Calling FAC. In addition, the route code function and implied precedence level are provided.

When WNDP is disabled, users must dial the Precedence Calling FAC, followed by the precedence level (0-4). The WNDP FACs can be administered, but you cannot use them.

Music-on-Hold

Use the Music-on-Hold feature to automatically provide music, silence, or tone to a caller that:

- Is on hold
- Is transferred
- Is parked
- Waits in a queue

Detailed description of Music-on-Hold

This section provides a detailed description of the Music-on-Hold feature.

Use the Music-on-Hold feature to automatically provide music, silence, or tone to a caller. [Table 56, Music, silence, and tone options](#), on page 863 shows the audio options that you can provide to a user when a call:

- Is on hold
- Is transferred
- Is parked
- Waits in a queue

Table 56: Music, silence, and tone options

Caller Status	Music	Silence	Tone
On hold	Yes	No	Yes
On a trunk call that is being transferred	Yes	Yes	No
Parked	Yes	No	Yes
Waits in a queue	Yes	No	Yes

NOTE:

If you use equipment that rebroadcasts music or other copyrighted materials, you might 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).

Hardware requirements for Music-on-Hold

The Music-on-Hold feature requires the following hardware:

- None

Administering Music-on-Hold

The following steps are part of the administration process for the Music-on-Hold feature:

- [Assigning music tones, music ports, and music for transferred trunks](#)
- [Defining a Class of Restriction for Music-on-Hold](#)
- [Connecting a music source to the server](#)
- [Assigning a source of music to a port](#)

This section describes:

- Any prerequisites for administering the Music-on-Hold feature
- The screens that you use to administer the Music-on-Hold feature
- Complete administration procedures for the Music-on-Hold feature

Prerequisites for administering Music-on-Hold

You must complete the following actions before you can administer the Music-on-Hold feature:

- None

Screens for administering Music-on-Hold

Screen name	Purpose	Fields
<i>Feature-Related System Parameters</i>	Assign music tones, music ports, and music for transferred trunks.	• Music/Tone On Hold • Music Port • Music (or Silence) on Transferred Trunk Calls
<i>Music Sources</i>	Assign a music source to a port.	All
<i>CPE Trunk Group</i>	Connect a music source to the server.	All
<i>Class of Restriction</i>	Define a Class of Restriction (COR) for Music-on-Hold.	Hear System Music on Hold

Assigning music tones, music ports, and music for transferred trunks

To assign music tones, music ports, and music for transferred trunks:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the **Feature-Related System Parameters** screen ([Figure 223, Feature-Related System Parameters screen](#), on page 865).

Figure 223: Feature-Related System Parameters screen

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/SIP 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?	

- 2 In the Music/Tone on Hold field, perform one of the following actions:
 - Type **music** if you want a caller who is on hold to hear music.
 - Type **tone** if you want a caller who is on hold to hear a tone.
 - Type **none** if you want a caller who is on hold to hear neither music nor a tone.

If the Tenant Partitioning field on the **Optional Features** screen is set to **y**, you cannot administer the Music/Tone on Hold field. If the Tenant Partitioning field on the **Optional Features** screen set to **y**, you must use the Music Sources screen to assign music to a port.

- 3 In the Port field, type the port number that provides Music-on-Hold access. [Table 57, Port field values](#), on page 866 shows how to construct a port number.

You must specify a port on a TN763 Auxiliary Trunk circuit pack, or any supported analog line circuit pack.

The system displays the Port field when you type **music** in the Music/Tone on Hold field.

Table 57: Port field values

Characters	Description	Value
1-2	Cabinet number	01 through 44 (For DEFINITY R configurations) 01 through 03 (For DEFINITY SI configurations) 01 through 64 (For S8700 IP-Connect)
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 SI, S8700 IP-Connect (tdm, pdm) configurations) 01 through 16 (ppp for S8700 IP-Connect) 01 through 08 (system-port for S8700 IP-Connect) 17/33 (ethernet on S8700 IP-Connect)
x	Administration without Hardware	If the Secondary data module? field, is set to n, you can type x in the Port field. A Port field set to x indicates that no hardware is associated with the port assignment.

- 4 In the **Music (or Silence) On Transferred Trunk Calls** field, perform one of the following actions:
 - Type **all** you want all transferred trunk calls to receive music until the call is answered.
 - Type **no** if you want a caller on the trunk call to hear:
 - Music while the caller waits to be transferred
 - Ringback tone as soon as the transfer is complete

The caller on the trunk call hears neither music nor a tone if Music-on-Hold is not administered.
 - Type **call-wait** if you want a trunk call that transfers to a station, and then waits at the station, to hear music, if music is administered.

All other transferred trunk calls receive the ringback tone.
- 5 Press **Enter** to save your changes.

Defining a Class of Restriction for Music-on-Hold

- 1 Type **change cor n**, where n is the Class of Restriction (COR) to which you want to add Music-on-Hold. Press **Enter**.
The system displays the *Class of Restriction* screen ([Figure 224, Class of Restriction](#), on page 867).

Figure 224: Class of Restriction

change cor 1
Page 1 of 4

CLASS OF RESTRICTION

COR Number: 1

COR Description:

FRL: 7 Can Be Service Observed? n Can Be A Service Observer? n Time of Day Chart: 2 Priority Queuing? n Restriction Override: none Restricted Call List? y Access to MCT? y Group II Category For MFC: 7 Send ANI for MFE? n MF ANI Prefix: Hear System Music on Hold? y	APLT? y Calling Party Restriction: none Called Party Restriction: none Forced Entry of Account Codes? n Direct Agent Calling? n Facility Access Trunk Test? n Can Change Coverage? n Fully Restricted Service? n Add/Remove Agent Skills? n Automatic Charge Display? n PASTE (Display PBX Data on Phone)? n Can Be Picked Up By Directed Call Pickup? n Can Use Directed Call Pickup? n Group Controlled Restriction: inactive

- 2 In the Hear System Music on Hold field, perform one of the following actions:
 - Type **y** if you want Music-on-Hold to be activate at a telephone.
 - Type **n** if you do not want Music-on-Hold to be activate at a telephone.
- 3 Press **Enter** to save your changes.

Connecting a music source to the server

To connect a music source to the server:

- 1 Type **add trunk-group next**. Press **Enter**.
 You use the customer-premises equipment (CPE) trunks to connect a music source to the server.
 For more information on how to administer trunk groups, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Assigning a source of music to a port

To assign the sources of music to a port:

- 1 Type **change music-sources**. Press **Enter**.
 The system displays the **Music Sources** screen ([Figure 225, Music Sources screen](#), on page 868).

Figure 225: Music Sources screen

change music-sources				Page 1 of X
Source	Type	Music Sources 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			

- 2 In the **Type** field, perform one of the following actions:
 - If you want the user to hear music, type **music**.
 - If you want the user to hear the tone-on-hold tone, type **tone**.
You can specify **tone** for only one music source on the **Music Sources** screen.
 - If you want the user to hear neither music nor a tone, type **none**.
- 3 In the **Port** field, type the auxiliary trunk address or the analog port address of the music source. You cannot enter duplicate addresses in the **Port** field.
The system displays the **Port** field only if you typed **music** in the **Type** field.
[Table 57, Port field values](#), on page 866 shows you how to construct a port number.
- 4 In the **Description** field, type a maximum of 20 characters that describe the source of the music.
The system displays the **Description** field, only if you typed **music** or **tone** in the **Type** field.
- 5 Press **Enter** to save your changes.

Reports for Music-on-Hold

The following reports provide information about the Music-on-Hold feature:

- None

Considerations for Music-on-Hold

This section provides information about how the Music-on-Hold feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Music-on-Hold under all conditions. The following considerations apply to Music-on-Hold:

- If the **Tenant Partitioning** field on the **Optional Features** screen is set to **y**, you cannot administer the **Music/Tone on Hold** field on the **Feature-Related System Parameters** screen. If the **Tenant Partitioning** field on the **Optional Features** screen set to **y**, you must use the **Music Sources** screen to assign music to a port.
- Any number of calls can simultaneously connect to music.
- The system does not provide music to callers, in a multicaller connection, who are in a queue, on hold, or parked.

Interactions for Music-on-Hold

This section provides information about how the Music-on-Hold feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Music-on-Hold in any feature configuration.

- **Automatic Call Distribution (ACD)**
If you administer Music-on-Hold to provide music, the system provides the music after the ACD split delayed announcement.
- **Data Privacy and Data Restriction**
If a user or an attendant places a call that has either Data Privacy or Data Restriction activated on hold, the system withholds Music-on-Hold. The system withholds Music-on-Hold to prevent transmission of a musical tone that a connected data service might falsely interpret as a data transmission.
- **Hunting**
If you administer Music-on-Hold to provide music, the system provides the music after the Direct Departmental Calling (DDC) group or the uniform call distribution (UCD) group delayed announcement.

Night Service

Use the Night Service feature to direct incoming calls to other answering points at night.

Avaya Communication Manager provides the following Night Service capabilities:

- Hunt Group Night Service
- Night Console Service
- Night Station Service
- Trunk Answer from Any Station
- Trunk Group Night Service

Detailed description of Night Service

This section provides a detailed description of the Night Service feature.

Hunt Group Night Service

With Hunt Group Night Service, an attendant or a split supervisor can assign a hunt group or a 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 that are waiting) in the queue, are directed to the night console.

To activate and deactivate this feature, the attendant usually presses the **Night** button on the principal attendant console or a designated console.

Night Station Service

Night Station Service directs incoming calls for the attendant to designated extensions. To activate Night Station Service, attendants press 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. Calls do not queue for the attendant.

When Night Station Service is active, the system routes the incoming calls to the attendant as follows:

- Direct Inward Dialing (DID) Listed Directory Number (LDN) calls are routed to a designated DID-LDN night extension.

Night Service

Detailed description of Night Service

- 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 are routed 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 terminating extension group (TEG)).

Trunk Answer from Any Station (TAAS)

With Trunk Answer from Any Station (TAAS), telephone users can answer all incoming calls to the attendant when the attendant is not on duty, and when other telephones are not designated to answer the calls. The incoming call activates a gong, a bell, or a chime and a telephone user dials an access code to answer the call.

Users can activate TAAS if each of the following conditions is met:

- The attendant pressed the **Night** button on the primary console or a user (if Communication Manager has no attendant administered) 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

With Trunk Group Night Service, an attendant or a designated Night Service telephone user can 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. The system redirects the calls that come into the trunk group to the group's designated 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.

A user can also 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. The system redirects any incoming calls made on the trunk groups to their designated NSE for the trunk group. 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 telephone.

To activate Trunk Group Night Service, you press the individual Trunk Night Service buttons on the attendant console or on a telephone. You can assign Trunk Night Service buttons on more than one telephone.

Hardware requirements for Night Service

The Night Service feature requires the following hardware:

- None

Administering Night Service

The following steps are part of the administration process for the Night Service feature:

- [Setting up night station service to voice mail](#)
- [Setting up Night Console Service](#)
- [Setting up Night Station Service](#)
- [Setting up Trunk Answer from Any Station](#)
- [Setting up external alerting](#)
- [Setting up external alerting Night Service](#)
- [Setting up Night Service for trunk groups](#)
- [Setting up Night Service for hunt groups](#)

This section describes:

- Any prerequisites for administering the Night Service feature
- The screens that you use to administer the Night Service feature
- Complete administration procedures for the Night Service feature

Screens for administering Night Service

Screen name	Purpose	Fields
<i>Hunt Group</i>	Set Night Station Service to voice mail.	Group Number
<i>Listed Directory Number</i>		Night Destination
<i>Console Parameters</i>		DID-LDN Only to LDN Night Ext
<i>Attendant Console</i>	Set up Night Console Service.	Console Type
<i>Listed Directory Numbers</i>	Set up Night Station Service.	Night Destination
<i>Console Parameters</i>		DID-LDN Only to LDN Night Ext
<i>Feature Access Codes</i>	Set up Trunk Answer from Any Station (TAAS)	Trunk Answer Any Station Access Code

Screen name	Purpose	Fields
<i>Console Parameters</i>	Set up external alerting.	EXT Alert Port (TAAS)
<i>Listed Directory Numbers</i>	Set up external alerting night service.	Night Destination
<i>Console Parameters</i>		- EXT Alert Port (TAAS) - DID-LDN to Night Ext.
<i>Trunk Group</i>	Set up Trunk Group Night Service.	Night Service
<i>Hunt Group</i>	Set up Night Service for hunt groups.	Night Service Destination

Setting up night station service to voice mail

To set up a night station service to voice mail:

- 1 Type **add hunt-group next**. Press **Enter**.

The system displays the **Hunt Group** screen ([Figure 226, Hunt Group screen](#), on page 874).

Figure 226: Hunt Group screen

Page 1 of X

HUNT GROUP

Group Name: ldn nights

Group Number: 5

MM Early Answer? _

Queue? _

Security Code: ____

ISDN Caller Disp: _____

Measured: _____

Controlling Adjunct: ____

Multiple Call Handling: _____

Objective: ____

Queue Length: ____

Calls Warning Threshold: ____

Time Warning Threshold: ____

Redirect on No Answer (rings): _

Forced Entry of Stroke Counts or Call Work Codes? _

Group Extension: 51002__

Skill? _

Vector? _

Group Type:

ACD? _

AAS? _

COR: _

TN: _

Supervisor: Extension: _____

Calls Warning Port: __

Time Warning Port: __

The Group Number field fills in automatically with the next hunt group number.

- 2 In the Group Name field, type the name of the group.

This example uses **ldn nights** in the Group Name field. There can 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 is **change tenant *n***. If you are using tenant partitioning, the **Night Destination** field does not appear on the **Listed Directory Numbers** screen. Instead, the **Night Destination** field is on the **Tenant** screen.

- 4 Type **change listed-directory-numbers**. Press **Enter**.

The system displays the **Listed Directory Numbers** screen ([Figure 227, Listed Directory Numbers screen](#), on page 875).

Figure 227: Listed Directory Numbers screen

Page	1	of	2
LISTED DIRECTORY NUMBERS			
Night Destination: 51002			
	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

- 5 In the **Night Destination** field, add the night destination on the listed directory phone.

This example uses **51002**.

- 6 Press **Enter** to save your changes.

- 7 Type **change console-parameters**. Press **Enter**.

The system displays the **Console Parameters** screen ([Figure 228, Console Parameters screen](#), on page 876).

Figure 228: Console Parameters screen

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

- 8** In the DID-LDN Only to LDN Night Ext 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 the example shown here, dial 51002.

NOTE:

You should receive a confirmation tone that consists of three beeps. This step is very important because calls to the LDN night service extension do not follow coverage.

- 11** In your voice mail, build the automated attendant with the extension of the Listed Directory Number (LDN), not the hunt group.

The originally dialed number was the LDN, which is the number Communication Manager passes to the voice mail application. 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 automated attendant mailbox.

Setting up Night Console Service

To set up night console service:

- 1** Type **change attendant *n***, where ***n*** is the number of the attendant console. Press **Enter**.

The system displays the **Attendant Console** screen ([Figure 229, Attendant Console screen](#), on page 877).

Figure 229: Attendant Console screen

```

                                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**.

The system can include only one night-only or one day/night console, 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

To set up night station service:

- 1** Type **change listed-directory-numbers**. Press **Enter**.

The system displays the **Listed Directory Numbers** screen ([Figure 230, Listed Directory Numbers screen](#), on page 878).

Figure 230: Listed Directory Numbers screen

LISTED DIRECTORY NUMBERS		
Night Destination: 1234		
Ext	Name	TN
1: 8100	attendant 8100	
2:		
3:		
4:		
5:		
6:		
7:		
8:		
9:		
10:		

- 2 Type the extension number in the **Night Destination** field. In the example shown here, the extension number is **1234**.

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**. Press **Enter**.

The system displays the **Console Parameters** screen ([Console Parameters screen](#) on page 878).

Figure 231: Console Parameters screen

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
Attendant Vectoring VDN:		

- 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, you must have the attendant use the night console button to activate and deactivate night service.

Setting up Trunk Answer from Any Station

To set the FAC for TAAS:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the *Feature Access Code (FAC)* screen ([Figure 232, Feature Access Code \(FAC\) screen](#), on page 879).

Figure 232: Feature Access Code (FAC) screen

change feature-access-codes	FEATURE ACCESS CODE (FAC)	Page 3 of x
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:		
Station User Admin of FBI Assign:	Remove:	
Station User Button Ring Control 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: 71		
User Control Restrict Activation:	Deactivation:	
Voice Coverage Message Retrieval Access Code:		
Voice Principal Message Retrieval Access Code: *80		

- 2 In the Trunk Answer Any Station Access Code field, type the access code. In this example, the access code is **71**.
- 3 Press **Enter** to save your changes.

Setting up external alerting

Once you set the FAC, you must determine where the external alerting device is connected to the server that is running Communication Manager (this example uses port 01A0702).

To set up external alerting:

- 1 Type **change console-parameters**. Press **Enter**.

The system displays the *Console Parameters* screen ([Figure 233, Console Parameters screen](#), on page 880).

Figure 233: Console Parameters screen

change console-parameters	CONSOLE PARAMETERS	Page 1 of x
Attendant Group Name: OPERATOR		
COS: 0		COR: 0
Calls in Queue Warning: 5		Attendant Lockout? y
Ext Alert Port (TAAS): 01A0702		
CAS: none		
IAS (Branch)? n		Night Service Act. Ext.:
IAS Att. Access Code:		IAS Tie Trunk Group No.:
Backup Alerting? n		Alternate FRL Station:
Attendant Vectoring VDN:		DID-LDN Only to LDN Night Ext? n

- 2 In the **EXT Alert Port (TAAS)** field, type the port address that is assigned to the external alerting device. In this example, the number is **01A0702**.
- 3 Press **Enter** to save your changes.

Setting up external alerting Night Service

To send Listed Directory Number (LDN) calls to the attendant during the day and to the desk of a security guard at night:

- 1 Type **change listed-directory-numbers**. Press **Enter**.
The system displays the **Listed Directory Numbers** screen ([Figure 234, Listed Directory Numbers screen](#), on page 880).

Figure 234: Listed Directory Numbers screen

LISTED DIRECTORY NUMBERS		
Night Destination: 3000		
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

- 2 In the **Night Destination** field, verify that this field is blank.
- 3 Press **Enter** to save your changes.

- 4 Type **change console-parameters**. Press **Enter**.

The system displays the **Console Parameters** screen ([Figure 235, Console Parameters screen](#), on page 881).

Figure 235: Console Parameters screen

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	
Attendant Vectoring VDN:		

- 5 In the **EXT Alert Port (TAAS)** field, type the port address that is assigned to the external alerting device. In this example, the number is **01A0702**.

- 6 Press **Enter** to save your changes.

The system is in Night Service.

Any calls to extension 2000 now go to extension 3000, the desk of the security guard.

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**. Press **Enter**.

The system displays the **Console Parameters** screen ([Figure 236, Console Parameters screen](#), on page 881).

Figure 236: Console Parameters screen

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	
Attendant Vectoring VDN:		

- 2 In the **DID-LDN Only to Night Ext.** field, type y.

This allows only LDN calls to go to the Listed Directory Night Service Number Extension.

- 3 In the **Ext Alert Port (TAAS)** field, type the port address that is assigned to the external alerting device. In this example, the number is **01A070**.

- 4 Press **Enter** to save your changes.

The system is in night service.

Any Dialed Number Identification Service (DNIS) extension 2000 and any 0-seeking calls now go to the TAAS bell.

Setting up Night Service for trunk groups

To set up Night Service for trunk groups:

- 1 Type **change trunk-group *n***, where *n* is the number of a trunk group. Press **Enter**.

In this example, you direct night calls for trunk group 2 to extension 1245. The system displays the **Trunk Group** screen ([Figure 237, Trunk Group screen](#), on page 882).

Figure 237: Trunk Group screen

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: ____
Bit Rate: 1200	Trunk Gain: high	
Disconnect Supervision - In? y	Synchronization: ____	Duplex: ____
Out? n		
Answer Supervision Timeout: 10		Receive Answer Supervision? n

- 2 In the **Night Service** field, type the extension number that you want the calls to go to In this example, the extension number is 1245.

The destination can be a station extension, a recorded announcement extension, a vector directory number (VDN), a hunt group extension, a terminating extension group (TEG), or **attnd** if you want to direct the call to the attendant.

- 3 Press **Enter** to save your changes.

Setting up Night Service for hunt groups

To set up Night Service for a hunt group:

- 1 Type **change hunt-group *n***, where *n* is the number of a hunt group. Press **Enter**.

The system displays the **Hunt Group** screen ([Figure 238, Hunt Group screen](#), on page 883).

Figure 238: Hunt Group screen

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 the extension number. In this example, the extension number is **1234**.

The destination can be an extension, a recorded announcement extension, a VDN, a hunt group extension, **attnd** if you want to direct calls to the attendant.

Calls to hunt group 3 follow the coverage path that is assigned to extension 1234.

- 3 Press **Enter** to save your changes.

Reports for Night Service

The following reports provide information about the Night Service feature:

- None

Considerations for Night Service

This section provides information about how the Night Service feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Night Service under all conditions. The following considerations apply to Night Service:

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 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 telephone with a trunk-ns button might 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 for Night Service

This section provides information about how the Night Service feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Night Service in any feature configuration.

Interactions for Hunt Group Night Service

- Automatic Call Distribution (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 using 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 LDN night destination, 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 by way of 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 by way of 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 LDN night destination, 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 that terminate 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.

Off-Premises Station

Use the Off-Premises Station feature to connect a telephone, that is in a different building than the server that runs Avaya Communication Manager, to your system.

Detailed description of Off-Premises Station

This section provides a detailed description of the Off-Premises Station feature.

With the Off-Premises Station feature, you can connect a telephone that is located outside the building where the server that runs Avaya Communication Manager to your system.

If you use central office (CO) trunk circuits, the telephone must be:

- Analog
- FCC-registered, if the telephone is in the United States
- Registered by the appropriate governmental agency, if the telephone is located outside the United States

You can use digital communications protocol (DCP) sets as off-premises telephones if you add the DEFINITY extender. You can set up IP stations as off-premise stations if you use point-to-point protocol (PPP) connections. DS1 trunk service provides a digital interface for off-premises stations.

A trunk-data module connects off-premises private-line trunk facilities and Avaya Communication Manager. The trunk-data module converts between the RS-232C and the DCP, and can connect to Direct Distance Dialing (DDD) modems as the DCP member of a modem pool.

Off-Premises Station requires cross-connecting capabilities, and one port on an analog line or a DS1 tie trunk circuit pack for each interface that you want to provide. Not all analog lines can support an off-premises station. For more information, see the user guide for the appropriate telephone.

The maximum loop distance for off-premises stations is 20,000 feet (6093.34 meters) if you do not use repeaters. For information about the cable distance, see the user guide for the appropriate telephone.

Note that the system does not support the use of a message waiting indicator lamp on an off-premises station.

Hardware requirements for Off-Premises Station

The Off-Premises Station feature requires the following hardware:

- None

Administering Off-Premises Station

The following steps are part of the administration process for the Off-Premises Station feature:

- [Activating Off-Premises Station for a user](#)

This section describes:

- Any prerequisites for administering the Off-Premises Station feature
- The screens that you use to administer the Off-Premises Station feature
- Complete administration procedures for the Off-Premises Station feature

Prerequisites for administering Off-Premises Station

You must complete the following actions before you can administer the Off-Premises Station feature:

- None

Screens for administering Off-Premises Station

Screen name	Purpose	Fields
Station (analog)	Activate an off-premises station for a user.	• Off Premise Station • R Balance Network

Activating Off-Premises Station for a user

To activate Off-Premises Station for a user:

- 1 Type **change station *n***, where *n* is the extension of the user for whom you want to activate an off-premises station. Press **Enter**.

The system displays the **Station** screen for the extension that you requested ([Figure 239, Station screen](#), on page 891).

Figure 239: Station screen

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?	
Model:		
Survivable GK Node Name:	Media Complex Ext:	
	IP Softphone? y	
	Remote Office Phone? y	

- 2 Page through the screens until you see the **Off-Premises Station** field.
- 3 In the **Off-Premises Station** field, perform the following actions:
 - Type **y** if telephone associated with this **Station** screen is not located in the same building as the server.
If you type **y**, you must administer the **R Balance Network** field on the **Station** screen.
 - Type **n** if the telephone associated **Station** screen is not located in the same building as the server.
- 4 In the **R Balance Network** field perform the following actions.
 - Type **y** to select the R Balance Capacitor network. In all other cases, except for those listed under **n**, type **y**.
 - Type **n**:
 - To select the standard resistor capacitor network
 - When the station port circuit is connected to the terminal equipment, for example, SLC carriers or impedance compensators, that are optioned for 600-ohm input impedance and the distance between the server and the equipment is less than 3,000 feet (914.4 meters).

You must complete the **R Balance Network** field if the **Off-Premises Station** field on the **Station** screen is set to **y**

Reports for Off-Premises Station

The following reports provide information about the Off-Premises Station feature:

- None

Considerations for Off-Premises Station

This section provides information about how the Off-Premises Station feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Off-Premises Station under all conditions. The following considerations apply to Off-Premises Station:

- None

Interactions for Off-Premises Station

This section provides information about how the Off-Premises Station feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Off-Premises Station in any feature configuration.

- Distinctive Ringing

The Distinctive Ringing feature might function improperly at an off-premises telephone because of the distance of the telephone from the server. However, the Distinctive Ringing feature can be disabled when you set the **Off-Premises Station** field on the **Station** screen to **y**. If the Distinctive Ringing feature is not used with an off-premises station, the telephone receives one-burst ringing for all calls.

Personal Station Access

Use the Personal Station Access (PSA) feature to allow a user to associate the preferences and permissions of the user telephone with another telephone of the same type.

Detailed description of Personal Station Access

This section provides a detailed description of the Personal Station Access (PSA) feature.

With the PSA feature, users can associate the preferences and the permissions of the user telephone with another telephone of the same type.

Telecommuting

With PSA, different users can use the same group of telephones at different times. For example, several telecommuting users can use the same office on different days of the week. The users use PSA to associate with the office telephone. When a user associates the telephone, the telephone is assigned to that user. For example, the user can originate and receive calls.

At home, a telecommuting can also use PSA to install a digital communications protocol (DCP) telephone and a DEFINITY Extender. The user can then all into the system, and use PSA to associate the home telephone with the that is extension assigned to the user at the office. When someone calls the user extension, the call rings at the home of the user.

When a user no longer wants to use PSA, the user disassociates from the telephone.

PSA requires a user to enter a security code, from either an on-site or an off-site telephone.

Invalid attempts to use PSA

Invalid attempts to use PSA generate referral calls, If the Security Violation Notification (SVN) feature is enable, the SVN software logs the invalid attempt. If a user hangs up, or presses the release button, the system does not log the action as an invalid attempt.

Preferences and permissions

The preferences and the permissions that are assigned to the user extension, and that are retained with PSA include:

- the definition of terminal buttons
- abbreviated dial lists
- Class of Service (COS) permissions
- Class of Restriction (COR) permissions

Extensions that do not have a COS, such as expert agent selection (EAS) agents or hunt groups, cannot use PSA.

Button mapping

PSA functions only on analog, hybrid, and digital communications protocol (DCP) telephones. Many types of DCP telephones exist, with different types and numbers of buttons. If you attempt to associate a DCP telephone or a hybrid telephone with a telephone that has incompatible buttons, button mapping is unpredictable. Telephones and ports on different media servers or switches cannot be associated through PSA. Telephones on different switches, or nodes, within Distributed Communications Systems (DCS) cannot be associated through PSA. The system does not limit the number of stations that can use PSA. However, heavy use of the associate and dissociate functions can have a temporary impact on system performance.

Unanswered calls

When a call goes to coverage from a PSA-disassociated extension, the software sends a message to the coverage point to indicate that the call was unanswered. If the coverage point is a display telephone, the system displays the “a,” which means “don’t answer.” If the coverage point is a voice messaging system, the voice messaging system receives an indication from the software that the call was unanswered, and the voice messaging system processes the call as unanswered.

Dissociated telephones

When a user requests to associate a telephone with PSA, any other telephone that uses the extension is automatically dissociated. Users can place emergency calls from a dissociated telephone, if a COR is assigned to dissociated telephones on the *Feature-Related System Parameters* screen.

PSA allows a dissociate request from a bridged appearance. However, the system dissociates the telephone from which the user issues the PSA command, even if the user is on a bridged appearance of another telephone.

The dissociate function within PSA allows a user to restrict the features that are available at a telephone. When a user uses PSA to dissociate a telephone, the telephone can only be used to:

- Call an attendant
- Accept a terminal translation initialization (TTI) or a PSA request

To enable users to make other types of calls from dissociated sets, you must establish a COR for the telephones.

Security



SECURITY ALERT:

Once an extension has been associated with a telephone, users of that telephone have the capabilities that are associated with the extension. You must issue a dissociate command from the telephone to ensure that unauthorized users cannot use the telephone. Dissociate the telephones if you use PSA and DCP extenders to permit remote DCP access.

Hardware requirements for Personal Station Access

The Personal Station Access feature requires the following hardware:

- None

Administering Personal Station Access

This section contains the screens that you use to administer the Personal Station Access (PSA) feature.

Screens for administering Personal Station Access

Screen name	Purpose	Fields
<i>Class of Service</i>	Assign PSA to a Class of Service (COS).	Personal Station Access
<i>Feature Access Code (FAC)</i>	Define a feature access code (FAC) to associate and dissociate telephones.	Personal Station Access (PSA) Associate Code and Dissociate Code
<i>Feature-Related System Parameters</i>	Specify the Class of Restriction (COR) to apply to calls made from dissociated telephones.	COR for PSA Dissociated Sets
	Specify the calling party number or automatic number identification (ANI) for calls that are made from dissociated telephones.	CPN, ANI for PSA Dissociated Sets
	Specify that the system record PSA transactions in the history log.	Record CTA/PSA/TTI Transactions in History Log?
<i>Station</i>	Define the station security code (SSC) for the extension.	Security Code

End-user procedures for Personal Station Access

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Personal Station Access (PSA):

- Using the PSA associate command
- Interrupting the PSA command sequence
- Using the PSA dissociate command

Using the PSA associate command

To use the PSA associate command, perform the actions shown in [Table 58, PSA associate commands](#), on page 896.

Table 58: PSA associate commands

Step	User action	System response	Result
1	Enter the feature access code (FAC) for PSA associate.	Dial tone	The request is successful.
		Intercept tone	• The request is unsuccessful, because the request was not made at an analog, hybrid, or DCP telephone. • The request is unsuccessful, because the telephone is associated with or assigned to a station that does not have PSA permission in its COS.
2	Enter an extension, and then a pound sign (#).	No response	If the user is at a telephone with a display, this pound sign (#) is the last character that the system displays until the PSA sequence is complete.

1 of 2

Table 58: PSA associate commands

Step	User action	System response	Result
3	Enter the Station Security Code (SSC) for the user extension, and then a pound sign (#).	Confirmation tone	The command sequence is successfully complete. The system queues the request.
		Intercept tone immediately after the confirmation tone.	The request is unsuccessful, because: • Terminal translation initialization (TTI) is not enabled for voice. • Either the dialed extension, or the originating extension, has an add, change, or remove action in progress. • You entered more that 15 digits before you entered the pound sign (#). Issue the PSA request again.
		Intercept tone	The request is unsuccessful because: • The extension and the SSC are incompatible. — The extension is invalid. — The SSC is invalid for the extension. — Both the extension and the SCC are invalid. The Security Violation Notification (SVN) feature logs this unsuccessful use of PSA as an invalid attempt. • The Class of Service (COS) of the extension does not allow PSA. • An extension in one tenant partition cannot be associated with a telephone in another tenant partition.
		Reorder tone	The request is unsuccessful because: • The extension that was entered is in use. • An agent is logged in at the dialed extension. • The system load is too heavy to allow the request to occur. The user can try again later.

2 of 2

Interrupting the PSA associate command sequence

If you enter incorrect information after you enter the FAC, you can interrupt the command sequence and begin again. To interrupt the command sequence:

- 1 Enter an asterisk (*) at any time before you enter the second pound sign (#).
The system generates dial tone
- 2 Enter the extension, again, but do not enter the FAC again.

SVN does not record the interrupted command sequence as an invalid attempt.

Using the PSA dissociate command

To use the PSA dissociate command,:

- Enter the (FAC for PSA dissociate.

The system responds with:

- Confirmation tone if the:
 - System successfully dissociates the telephone
 - Telephone was not previously associated
- Intercept tone, if the COS of the telephone extension does not allow the use of PSA.

Reports for Personal Station Access

The following reports provide information about the Personal Station Access feature:

- None

Considerations for Personal Station Access

This section provides information about how the Personal Station Access (PSA) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Personal Station Access under all conditions. The following considerations apply to Personal Station Access:

- None

Interactions for Personal Station Access

This section provides information about how the Personal Station Access (PSA) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Personal Station Access in any feature configuration.

- Adjunct/Switch Application Interface (ASAI)

You cannot use PSA on an ASAI link, because ASAI uses a Basic Rate Interface (BRI) port. Do not assign a Class of Service (COS) that allows PSA to an ASAI link.

- Bridged Appearance

When you issue a PSA dissociate request for the principal telephone, the bridged appearances of the telephone remain active. The bridged appearances remain active if the telephones upon which the bridged appearances appear have not been dissociated.

When a call is made to the principal extension, any of the bridged appearances can alert. If the call cannot alert at a bridged appearance of the principal extension, the system routes the call to the coverage path of the principal extension.

PSA allows a dissociate request from a bridged appearance. However, the system dissociates the telephone at which the user issues the PSA command, even if the user is on a bridged appearance of another telephone.

- Call Management

A PSA dissociate request automatically logs out an Automatic Call Distribution (ACD) agent.

- Coverage

PSA does not change coverage path operations. If a station is dissociated, the system routes calls to coverage, unless the calls are forwarded.

- Property Management System (PMS)

Do not assign a COS that allows PSA to an extension that is assigned to a room, instead of to a user.

- Save Translations

PSA commands cannot be successfully executed during a save translations operation. When a reset 3 or greater (reset 4, reset 5, and so on) occurs on the system, all associations revert to the state as of the last save translations operation.

- Security Violation Notification (SVN)

If SVN is active, SVN tracks and reports PSA security violations.

- Tenant Partitioning

If a telephone is already associated, a user who attempt a PSA associate request at that telephone must specify an extension that is in the same partition as the extension that is already associated with the telephone.

However, any user, in any partition, can issue a PSA dissociate request at the telephone, if the COS of the telephone extension allows PSA. After the user successfully dissociates the extension, the user can issue a PSA associate request for an extension in any tenant partition.

Personalized Ringing

Use the Personalized Ringing feature to hear one of eight ringing patterns for incoming calls. You assign the ringing pattern to each user on your system. The different ringing patterns help users who work in the same area to distinguish their calls from the calls of other users.

Detailed description of Personalized Ringing

This section provides a detailed description of the Personalized Ringing feature.

You can administer Personalized Ringing for each telephone. Either you, or the end user, can administer Personalized Ringing for some programmable telephones.

Ringing patterns

The eight administrable ringing patterns are different combinations of three tones. The eight tone combinations are (in Hertz):

- 750, 750, 750 (standard ringing)
- 1060, 1060, 1060
- 530, 530, 530
- 530, 1060, 1060
- 1060, 1060, 530
- 1060, 530, 530
- 1060, 530, 1060
- 530, 1060, 530

Power failures

The user-specified ringing pattern for some digital voice terminals is lost when the power fails. The system retains the user-specified ringing pattern for ISDN-BRI voice terminals when the power fails.

Hardware requirements for Personalized Ringing

The Personalized Ringing feature requires the following hardware:

- None

Administering Personalized Ringing

The following steps are part of the administration process for the Personalized Ringing feature:

- [Assigning Personalized Ringing to a user telephone](#)

This section describes:

- Any prerequisites for administering the Personalized Ringing feature
- The screens that you use to administer the Personalized Ringing feature
- Complete administration procedures for the Personalized Ringing feature

Prerequisites for administering Personalized Ringing

You must complete the following actions before you can administer the Personalized Ringing feature:

- None

Screens for administering Personalized Ringing

Screen name	Purpose	Fields
<i>Station</i>	Assign Personalized Ringing to a user telephone.	Personalized Ringing Pattern

Assigning Personalized Ringing to a user telephone

To assign Personalized Ringing to a user telephone:

- 1 Type **change station *n***, where *n* is the user telephone to which you want to assign Personalized Ringing. Press **Enter**.

The system displays the *Station* screen ([Figure 240, Station screen](#), on page 903).

Figure 240: Station screen

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?	
Model:		
Survivable GK Node Name:	Media Complex Ext:	
	IP Softphone? y	
	Remote Office Phone? y	

- 2 In the Personalized Ringing Pattern field, type one of the valid entries shown in [Table 59, Personalized ringing patterns](#), on page 903.

Use the following key to the Usage column:

- L = 530 Hz
- M = 750 Hz
- H = 1060 Hz

For virtual stations, the Personalized Ringing Pattern field dictates the ringing pattern on the mapped-to physical telephone.

Table 59: Personalized ringing patterns

Valid entries	Usage
1	MMM (standard ringing)
2	HHH
3	LLL
4	LHH
5	HHL
6	HLL
7	HLH
8	LHL

- 3 Press **Enter** to save your changes.

Reports for Personalized Ringing

The following reports provide information about the Personalized Ringing feature:

- None

Considerations for Personalized Ringing

This section provides information about how the Personalized Ringing feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Personalized Ringing under all conditions. The following considerations apply to Personalized Ringing:

- None

Interactions for Personalized Ringing

This section provides information about how the Personalized Ringing feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Personalized Ringing in any feature configuration.

- Distinctive Ringing

With Distinctive Ringing, you can administer the relationship between the number of ring bursts and the call type. The Personal Ringing Pattern that you select is the same ringing pattern used in the Distinctive Ringing cycles.

Posted Messages

Use the Posted Messages feature to provide callers with a displayed message on the telephone that states why the user is unavailable to take a call.

Detailed description of Posted Messages

This section provides a detailed description of the Posted Messages feature.

The Posted Messages feature is available with Avaya Communication Manager release 1.3 (V11) or later.

A user activates the Posted Messages feature to post a message to his or her telephone. When a person calls that user, the telephone display of the caller shows the posted message for 4 seconds. During those 4 seconds, no incoming alert can appear on the telephone of the caller. After 4 seconds, the telephone display of the caller reverts back to the normal called number and called name.

The Posted Messages feature assumes that the telephone of the caller can display messages. Only internal callers can post or view a posted message. Callers from outside your system cannot view a posted message.

Language options

Five languages are available for the Posted Messages feature:

- English
- Italian
- French
- Spanish
- A user-defined language

The 15 fixed messages and feature button labels are available in English, with predefined Italian, French, and Spanish translations. The administrator cannot change the text of the English, the Italian, the French, or the Spanish fixed messages or feature button labels.

The administrator can translate the fixed messages and feature button labels into a user-defined language. This translation can be any other language that the customer chooses, such as German. The administrator can use only one user-defined language throughout the system.

The 15 custom messages can be administered in English, Italian, French, Spanish, and the user-defined language. The number of available messages equals the number of fixed messages (15), plus the number of custom messages that you administer for each language.

Language translation is automatically achieved between telephones and between systems. Let us say that telephone A uses English and telephone B uses Italian. User A posts the message "In Meeting." User B calls user A and sees the message "In Riunione" in Italian. If properly administered, the same is true for custom messages.

Messages

User can choose from up to 30 different messages. Of these 30 messages, 15 are *fixed* messages, and 15 are *custom* messages. Each message has a corresponding message number.

While a user selects a Posted Message, his or her telephone is said to be in *selection display mode*. Once the user posts a message, his or her telephone is said to be in *message posting mode*. The message appears on the telephone display of the user. The display of the selected message on the telephone is a visual reminder to the user. The display also indicates the availability of the user to people who might walk into the office.

Telephones that are in message posting mode use a special dial tone when the telephone goes off hook. This special dial tone provides audio feedback to the user. On telephones without a display, this special dial tone is the only indication that the telephone is in message posting mode.

The telephone of the user displays the selected message until:

- someone deactivates the message
The user or someone else can deactivate the posted message on the telephone of the user.
- the system resets
If the system resets, the posted message no longer appears on the telephone of the user.

Fixed messages

The numbers for the fixed messages are predefined. You cannot change the numbers or the messages. The following table shows the 15 fixed messages.

Number	Message
01	In Meeting
02	Out To Lunch
03	Away From Desk
04	Do Not Disturb
05	Out All Day
06	On Vacation
07	Gone For The Day
08	Out Sick
09	On Business Trip
10	With Client
11	Working From Home
12	On Leave
13	Back In 5 Minutes

Number	Message
14	Back In 30 Minutes
15	Back In 1 Hour

Note that the Do Not Disturb posted message is independent of the Do Not Disturb feature. Activating or deactivating the Do Not Disturb posted message has no impact on the Do Not Disturb feature.

Custom messages

If you choose to add custom messages, you must start with number 16 and continue in numeric order. Custom messages are specific to each local system. Custom messages in any language cannot exceed 28 characters.

QSIG support

If your system supports QSIG functionality, you can use QSIG to send fixed messages and custom messages to users on other systems. QSIG is a global signaling and control standard for use in private corporate ISDN networks.

The 15 fixed messages are automatically routed to other systems through QSIG. If you want to send custom messages to users on other systems through QSIG, you must:

- Activate QSIG support
- Create the same custom messages on all systems.

If you do not administer custom messages on the system of the caller, a message does not appear on the telephone display of the caller.

Custom messages must be consistent for all systems and for all translations. The reason is that Posted Messages sends the message number, not the message, to the caller. The system of the caller converts the Posted Messages number into a message.

Let us say that custom message #16 on system A is “On Conference Call” and custom message #16 on system B is “Talking to Boss.” If a user on system A posts custom message #16, “On Conference Call” appears on the telephone display of the user. If someone on system B calls the user on system A, Posted Messages sends message number 16 to the caller. The system of the caller converts number 16 into a message. “Talking to Boss” appears on the telephone display of the caller.

Inconsistent administration on custom messages can display unintended results. The system might not display the intended message.

Hardware requirements for Posted Messages

The Posted Messages feature requires the following hardware:

- A display telephone for the caller

Users do not need a display telephone to post a message.

- A user with a display or nondisplay enhanced tip ring (ETR) telephone can post a message, but ETR display telephones do not show posted messages.

Administering Posted Messages

The following steps are part of the administration process for the Posted Messages feature:

- [Defining a feature access code \(FAC\)](#)
- [Requiring a security code](#)
- [Activating QSIG to send custom messages](#)
- [Translating Posted Messages](#)
- [Translating telephone feature buttons and labels](#)

This section describes:

- Any prerequisites for administering the Posted Messages feature
- The screens that you use to administer the Posted Messages feature
- Complete administration procedures for the Posted Messages feature

Prerequisites for administering Posted Messages

You must complete the following actions before you can administer the Posted Messages feature:

- On the **Optional Features** screen, ensure that the G3 **Version** field is set to **V11** or later. If this field is not set to **V11** or later, your system is not enabled for the Posted Messages feature. Contact your Avaya representative for assistance.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

- Click **Next** until you see the ISDN-BRI Trunks and ISDN-PRI fields. Ensure that at least one of these fields are set to **y**, depending on what type of trunk you use. These fields indicate your system can support QSIG to send and receive Posted Messages. Your license file sets the values in the ISDN-BRI Trunks and ISDN-PRI fields. You cannot manually change these values. If both fields are set to **n**, see your Avaya representative for assistance.
- Click **Next** until you see the Posted Messages field. Ensure that the Posted Messages field is set to **y**. Your license file sets the value in the Posted Messages field. You cannot manually change this value. If the Posted Messages field on the **Optional Features** screen is set to **n**, see your Avaya representative for assistance.

- Click **Next** until you see the **QSIG Optional Features** screen. Ensure that the following fields are set to **y**:

- Basic Call Setup
- Basic Supplementary Services
- Value-Added (VALU)

These fields indicate whether your system can support QSIG to send and receive Posted Messages. Your license file sets the values in these fields. You cannot manually change these values. If any of these fields are set to **n**, see your Avaya representative for assistance.

For a complete description of the **Optional Features** screens, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Posted Messages

Screen name	Purpose	Fields
Optional Features	Ensure that you have Communication Manager version 1.3 (V11) or later.	G3 Version
	Ensure that your system supports QSIG functionality.	• ISDN-BRI Trunks • ISDN-PRI
	Ensure that the Posted Messages feature is on.	Posted Messages
QSIG Optional Features	Ensure that your system supports QSIG functionality.	• Basic Call Setup • Basic Supplementary Services • Value-Added (VALU)
Feature Access Code (FAC)	Set up a FAC for users to activate and deactivate the Posted Messages feature.	• Posted Messages Activation • Deactivation
Feature-Related System Parameters	Activate QSIG to send custom messages to people on other systems.	Send Custom Messages Through QSIG
	Indicate if users must enter a security code to activate or deactivate Posted Messages using a FAC.	Require Security Code
Station	Set up a security code for the telephone of a user.	Security Code

Screen name	Purpose	Fields
<i>System Posted Messages</i>	Review the 15 fixed messages and, if needed, translate the messages to a user-defined language.	All
<i>Custom Posted Messages</i>	Create or edit up to 15 custom messages in English, Italian, French, Spanish, or a user-defined language.	All
<i>Language Translations</i>	If needed, change the translation of the Posted Messages feature display to a user-defined language.	Posted Messages
	If needed, change the translation of the Posted Messages softkey label to a user-defined language.	PoMsg
	If needed, change the translation of the Posted Messages button label on the 2420 telephone or the 4620 telephone to a user-defined language.	Posted MSGs

Defining a feature access code (FAC)

To define the two feature access codes (FAC) to activate and deactivate the Posted Messages feature:

- 1 Type **change feature-access-codes**. Press **Enter**.
The system displays the **Feature Access Code (FAC)** screen.
- 2 Click **Next** until you see the Posted Messages Activation field ([Figure 241, Feature Access Code \(FAC\) screen](#), on page 911).

Figure 241: Feature Access Code (FAC) screen

change feature-access-codes	Page 3 of 8
FEATURE ACCESS CODE (FAC)	
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:	
Posted Messages Activation: *29	Deactivation: #29
Priority Calling Access Code:	
Program Access Code:	
Refresh Terminal Parameters Access Code:	
Remote Send All Calls Activation:	Deactivation:
Self Station Display Activation:	
Send All Calls Activation:	Deactivation:
Station Firmware Download Access Code:	
Station Lock Activation:	Deactivation:
Station Security Code Change Access Code:	
Station User Admin of FBI Assign:	Remove:
Station User Button Ring Control Access Code:	
Terminal Dial-Up Test Access Code:	

- 3 In the Posted Messages Activation field, type a FAC to activate Posted Messages.
- 4 In the Deactivation field, type a FAC to deactivate Posted Messages.
- 5 Press **Enter** to save your changes.

Requiring a security code

An administrator can require a security code when users activate or deactivate Posted Messages using a FAC. The requirement for the security code applies to all users. This procedure is optional.

To require users to enter a security code when using a FAC to activate or deactivate Posted Messages, you must complete the following procedures:

- [Setting up a security code for the telephone of a user](#)
- [Activating the Posted Messages security code](#)

Setting up a security code for the telephone of a user

To set up a security code for the telephone of a user:

- 1 Type **change station *n***, where *n* is the telephone extension of the user. Press **Enter**.
The system displays the **Station** screen ([Figure 242, Station screen](#), on page 912).

Figure 242: Station screen

change station 1234567 Page 1 of 3

STATION

Extension: 1234567	Lock Messages? n	BCC: 0
Type: 4606	Security Code: 86562563	TN: 1
Port: S00014	Coverage Path 1:	COR: 1
Name: MG IP phone 4606	Coverage Path 2:	COS: 1
	Hunt-to Station:	

STATION OPTIONS

Loss Group: 19	Personalized Ringing Pattern: 1
Speakerphone: 2-way	Message Lamp Ext: 1234567
Display Language: english	Mute Button Enabled: y

- 2 In the **Security Code** field, type a security code for this telephone.
The security code can be up to eight digits.
- 3 Press **Enter** to save your changes.
- 4 Be sure to share this security code with the user.

Activating the Posted Messages security code

- 1 Type **change system-parameters features**. Press **Enter**.
The system displays the *Feature-Related System Parameters* screen.
- 2 Click **Next** until you see the Posted Message area ([Figure 243, Feature-Related System Parameters screen](#), on page 913).

Figure 243: Feature-Related System Parameters screen

change system-parameters features Page 6 of 12

FEATURE-RELATED SYSTEM PARAMETERS

CONFERENCE/TRANSFER

Abort Transfer? n
Transfer Upon Hang-Up? n
Abort Conference Upon Hang-Up? n
No Hold Conference Timeout: 60

No Dial Tone Conferencing? n
Select Line Appearance Conferencing? n
Unhold? n

ANALOG BUSY AUTO CALLBACK
Without Flash? n

AUDIX ONE-STEP RECORDING
Recording Delay Timer (msec): 500
Apply Ready Indication Tone To Which Parties In The Call? all
Interval For Applying Periodic Alerting Tone (seconds): 15

POSTED MESSAGE
Require Security Code? y

- 3 Change the Require Security Code field to y.
This field appears only if the Posted Messages field on the *Optional Features* screen is set to y.
- 4 Press **Enter** to save your changes.

Activating QSIG to send custom messages

To activate QSIG to send custom messages to users on other systems:

- 1 Type **change system-parameters features**. Press **Enter**.
The system displays the *Feature-Related System Parameters* screen.
- 2 Click **Next** until you see the ISDN Parameters area ([Figure 244, Feature-Related System Parameters screen](#), on page 914).

Figure 244: Feature-Related System Parameters screen

change system-parameters features	Page7 of 12
FEATURE-RELATED SYSTEM PARAMETERS	
ISDN PARAMETERS	
Send Non-ISDN Trunk Group Name as Connected Name? n	
Display Connected Name/Number for ISDN DCS Calls? n	
Send ISDN Trunk Group Name on Tandem Calls? n	
Send Custom Messages Through QSIG? y	
QSIG TSC Extension: 6999	
MWI - Number of Digits Per Voice Mail Subscriber: 4	
National CPN Prefix:	
International CPN Prefix:	
Pass Prefixed CPN to ASAI? n	
Unknown Numbers Considered Internal for AUDIX? n	
USNI Calling Name for Outgoing Calls? n	
Path Replacement with Measurements? y	
QSIG Path Replacement Extension:	
Path Replace While in Queue/Vectoring? n	

- 3 Change the Send Custom Messages Through QSIG field to y.
- 4 Press **Enter** to save your changes.

Translating Posted Messages

Translating any fixed or custom message is optional.

To translate Posted Messages, complete the following procedures:

- [Translating fixed messages to a user-defined language](#)
- [Creating custom messages](#)

Translating fixed messages to a user-defined language

Fixed messages are available in English, Italian, French, and Spanish. If you want to translate fixed messages into another language, called a user-defined language, follow this procedure.

To translate fixed messages to a user-defined language:

- 1 Type **change display-messages posted-message**. Press **Enter**.
The system displays the **System Posted Messages** screen ([Figure 245, System Posted Messages screen](#), on page 915).

Figure 245: System Posted Messages screen

change display-messages posted-message

Page 1 of 6

SYSTEM POSTED MESSAGES

USER DEFINED LANGUAGE TRANSLATIONS

Message Number	English	Message Number	Translation
1.	In Meeting	1.	*****
2.	Out to Lunch	2.	*****
3.	Away from Desk	3.	*****
4.	Do Not Disturb	4.	*****
5.	Out All Day	5.	*****
6.	On Vacation	6.	*****
7.	Gone for the Day	7.	*****
8.	Out Sick	8.	*****
9.	On Business Trip	9.	*****
10.	With Client	10.	*****
11.	Working from Home	11.	*****
12.	On Leave	12.	*****
13.	Back in 5 Minutes	13.	*****
14.	Back in 30 Minutes	14.	*****
15.	Back in 1 Hour	15.	*****

- 2 In the Translation fields, type a translation of all fixed messages into the user-defined language.

Note that the text for the English, the Italian, the French, and the Spanish fixed messages are predefined. You cannot change the text of these translations.

- 3 Press **Enter** to save your changes.

Creating custom messages

To create custom messages:

- 1 Type **change display-messages posted-message**. Press **Enter**.

The system displays the *System Posted Messages* screen.

- 2 Click **Next**.

The system displays the *Custom Posted Messages* screen for English translations ([Figure 246, Custom Posted Messages screen](#), on page 916).

The system displays the five *Custom Posted Messages* screens in the following order:

- English translations
- Italian translations
- French translations
- Spanish translations
- User-defined language translations

In this example, we describe how to create custom messages in English. The same instructions apply for creating custom messages in Italian, in French, in Spanish, or in the user-defined language. Ensure that you view the correct screen for the language that you want to use.

If you first administer custom messages in English, the text of the English custom messages automatically appears on the:

- Italian translations screen
- French translations screen
- Spanish translations screen
- user-defined language translations screen

Figure 246: Custom Posted Messages screen

change display-messages posted-message Page 2 of 6

CUSTOM POSTED MESSAGES
ENGLISH TRANSLATIONS

Message Number	Translation
16.	*****
17.	*****
18.	*****
19.	*****
20.	*****
21.	*****
22.	*****
23.	*****
24.	*****
25.	*****
26.	*****
27.	*****
28.	*****
29.	*****
30.	*****

- 3 In the Translation fields, start with message 16 and type the message in the proper language that you want to create.
- 4 Continue in numeric order until you create all the custom messages that you want. The maximum number of custom messages is 15.
- 5 Press **Enter** to save your changes.

Translating telephone feature buttons and labels

Translating telephone feature buttons and labels is optional.

To translate telephone feature buttons and labels for Posted Messages to a user-defined language, complete the following procedures:

- [Translating the Posted Messages feature display to a user-defined language](#)
- [Translating the Posted Messages softkey button label to a user-defined language](#)
- [Translating the Posted Messages button label to a user-defined language](#)

Translating the Posted Messages feature display to a user-defined language

To translate the text that appears on the telephone display, when this feature is active, to a user-defined language:

- 1 Type **change display-messages view-buttons**. Press **Enter**.
The system displays the *Language Translations* screen.
- 2 Click **Next** until you see the Posted Messages field ([Figure 247, Language Translations screen](#), on page 917).

Figure 247: Language Translations screen

change display-messages view-buttons		Page 9 of 9
LANGUAGE TRANSLATIONS		
English	Translation	
1. Station Lock	*****	
2. License Error	*****	
3. Conference Display	*****	
4. Conf/Trans Toggle-Swap	*****	
5. Posted Messages	*****	

- 3 In the Translation fields, type a translated name for Posted Messages into the user-defined language.
Note that the language translations for the English, the Italian, the French, and the Spanish feature display are predefined. You cannot change the text of these translations.
- 4 Press **Enter** to save your changes.

Translating the Posted Messages softkey button label to a user-defined language

To translate the Posted Messages softkey button label to a user-defined language:

- 1 Type **change display-messages softkey-labels**. Press **Enter**.
The system displays the *Language Translations* screen ([Figure 248, Language Translations screen](#), on page 918).

Figure 248: Language Translations screen

change display-messages softkey-labels				Page 9 of 9	
LANGUAGE TRANSLATIONS					
English	Translation	English	Translation	English	Translation
1. Acct	1. *****	17. Drop	17. *****	33. SAC	33. *****
2. AD	2. *****	18. Excl	18. *****	34. SFunc	34. *****
3. AdBsy	3. *****	19. GrpPg	19. *****	35. Spres	35. *****
4. Admin	4. *****	20. HFAns	20. *****	36. Stats	36. *****
5. AutCB	5. *****	21. IAuto	21. *****	37. Stop	37. *****
6. BtnVu	6. *****	22. IDial	22. *****	38. Swap	38. *****
7. CFrwd	7. *****	23. Inspt	23. *****	39. Timer	39. *****
8. CnfdDs	8. *****	24. Last	24. *****	40. TmDay	40. *****
9. CnLWC	9. *****	25. LWC	25. *****	41. View	41. *****
10. Cnslt	10. *****	26. Mark	26. *****	42. Wait	42. *****
11. Count	11. *****	27. Pause	27. *****	43. WspAn	43. *****
12. CPark	12. *****	28. PCall	28. *****	44. WspPg	44. *****
13. CPkUp	13. *****	29. PoMsg	29. *****		
14. CTime	14. *****	30. Prog	30. *****		
15. Dir	15. *****	31. RmBsy	31. *****		
16. DPkUp	16. *****	32. RngOf	32. *****		

- 2 Locate the PoMsg field.
- 3 In the Translation fields, type a translated name for the PoMsg softkey button label into the user-defined language. You are limited to five spaces for this translation.

Note that the language translations for the English, the Italian, the French, and the Spanish softkey button labels are predefined. You cannot change the text of these translations.
- 4 Press **Enter** to save your changes.

Translating the Posted Messages button label to a user-defined language

The 2420 DCP telephone and the 4620 IP telephone have digital button labels instead of paper labels. If you need to translate the Posted Messages button labels into a user-defined language for these telephones, follow this procedure.

To translate the Posted Messages button label for the 2420 DCP telephone and the 4620 IP telephone to a user-defined language:

- 1 Type **change display-messages button-labels**. Press **Enter**.
The system displays the *Language Translations* screen.
- 2 Click **Next** until you see the Posted MSGs field ([Figure 249, Language Translations screen](#), on page 919).

Figure 249: Language Translations screen

change display-messages button-labels		Page 9 of 9
LANGUAGE TRANSLATIONS		
English	Translation	
1. NoAnsAlrt	1.	*****
2. OffBoardAlarm	2.	*****
3. PMS Failure	3.	*****
4. PMS Ptr Alarm	4.	*****
5. Posted MSGs	5.	*****
6. Priority Call	6.	*****
7. QueueCall	7.	*****
8. QueueTime	8.	*****
9. Release	9.	*****
10. RemBusyInd	10.	*****
11. ResetAlert	11.	*****
12. Ringer Off	12.	*****
13. Ring Stat	13.	*****
14. RSVN Halt	14.	*****
15. SD	15.	**
16. SendAllCalls	16.	*****

- 3 In the Translation field, type a translated name for the Posted MSGs button label into the user-defined language.

Note that the language translations for the English, the Italian, the French, and the Spanish button labels are predefined. You cannot change the text of these translations.

- 4 Press **Enter** to save your changes.

End-user procedures for Posted Messages

End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Posted Messages:

To activate and deactivate the Posted Messages feature, users can either:

- dial a feature access code (FAC)
- press a feature button on the telephone

Users can dial the FAC from:

- their own telephone
- another telephone on the same system
- a remote access trunk

A user can dial the Posted Messages FAC from another telephone on the same system, and have the message post on their own telephone. Users must press the feature button on their own telephone to activate or deactivate the Posted Messages feature.

To post a message using a FAC, users may also have to dial the security code for their telephone. The administrator sets up the system to either require or not require a security code. Ask your administrator if you need to dial a security code, and what is the security code for your telephone.

Attendant consoles cannot use the Posted Messages feature. However, attendant consoles can use the Posted Messages FACs to activate or deactivate this feature for other telephones.

Activating the Posted Messages feature

To activate the Posted Messages feature, the user can complete the following procedures:

- [Using a feature access code \(FAC\)](#)
- [Using the Posted Messages feature button](#)

Using a feature access code (FAC)

To post a message with a feature access code (FAC):

- 1** Dial the Posted Messages activation FAC.
You hear dial tone.
- 2** Identify your extension as follows:
 - If you are dialing from your own telephone, just press the pound key (#).
 - If you are dialing from another internal telephone or a remote access trunk, dial your telephone extension. Then press the pound key (#).
- 3** Identify the security code for your telephone as follows:
 - If you do not have to dial a security code, just press the pound key (#).
 - If you do have to dial a security code, dial the security code for your telephone. Then press the pound key (#).

If there is an error, you hear intercept tone. Press the star key (*) to clear the extension and security code. Repeat from step 2. If you do not hear intercept tone, continue with step 4.
- 4** Dial the two-digit number of the message you want to post. Then press the pound key (#).
You hear confirmation tone. The system posts the message to the telephone display after one second as long as:
 - The telephone does not receive an incoming call
 - The user does not press any other button on the telephone

Examples

You want to post a message to your telephone. For the following examples:

- The activation FAC is ***29**.
- Your telephone extension is **1234567**.
- The security code for your telephone, if needed, is **86562563**.
- The message that you want to post to your telephone is Out To Lunch – message **02**.

From	Dial
Your own telephone requiring a security code,	*29#86562563#02#
Your own telephone not requiring a security code,	*29##02#
Another telephone requiring a security code,	*291234567#86562563#02#
Another telephone not requiring a security code,	*291234567##02#

Using the Posted Messages feature button

To post a message with the feature button on your telephone:

- 1** Press the Posted Messages feature button.
You can use this method to post a message only to your own telephone.
- 2** Select the message you want to post using one of these two ways:
 - Continually press the Posted Messages feature button to scroll through the available messages. Whenever you press the feature button, the next message appears. To select the message you want to post, press the pound key (#).
 - Dial the two-digit number of the message you want to post. Then press the pound key (#).

You hear confirmation tone. The system posts the message to the telephone display after one second as long as:

- The telephone does not receive an incoming call
- The user does not press any other button on the telephone

Deactivating the Posted Messages feature

To deactivate the Posted Messages feature, the user can complete the following procedures:

- [Using a feature access code \(FAC\)](#)
- [Using the Posted Messages feature button](#)

Using a feature access code (FAC)

To cancel the Posted Message feature with a feature access code (FAC):

- 1** Dial the Posted Messages deactivation FAC.
- 2** Identify your extension as follows:
 - If you are dialing from your own telephone, just press the pound key (#).
 - If you are dialing from another internal telephone or a remote access trunk, dial your telephone extension. Then press the pound key (#).

3 Identify the security code for your telephone as follows:

- If you do not have to dial a security code, just press the pound key (#).
- If you do have to dial a security code, dial the security code for your telephone. Then press the pound key (#).

If there is an error, you hear intercept tone. Press the star key (*) to clear the extension and security code. Repeat from step 2.

If there is no error, you hear confirmation tone. The telephone returns to the normal mode. The system clears the selected message.

Examples

You want to remove the posted message from your telephone. For the following examples:

- The deactivation FAC is **#29**.
- Your telephone extension is **1234567**.
- The security code for your telephone, if needed, is **86562563**.

From	Dial
Your own telephone requiring a security code,	#29#86562563#
Your own telephone not requiring a security code,	#29##
Another telephone requiring a security code,	#291234567#86562563#
Another telephone not requiring a security code,	#291234567##

Using the Posted Messages feature button

To cancel the Posted Message feature with the feature button on your telephone:

- 1** Press the Posted Messages feature button on your telephone.

You can use this method to cancel a posted message only on your own telephone.

You hear confirmation tone. The telephone returns to the normal mode. The system clears the selected message.

Reports for Posted Messages

The following reports provide information about the Posted Messages feature:

- None

Considerations for Posted Messages

This section provides information about how the Posted Messages feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Posted Messages under all conditions.

- Performance impact

Heavy use of the selection display mode might adversely affect system performance. Hardware buffers can overflow when you overuse telephone displays. Telephone hyperactivity checks are usually sufficient to control overuse of the display mode.

- Security

When a user activates the Posted Messages feature from a remote access trunk, the user must log in as a remote user. Logging in as a remote user prevents unauthorized users from activating or deactivating this feature for telephones.

- Serviceability

In the selection display mode, the telephone might go out of service if the user scrolls through the displays too quickly. Scrolling through the displays too quickly causes the telephone to be reset. Active calls are dropped. When a 2420, a 4620 or a 4630 telephone is out of service, the system clears the call log.

- Time out

If no activity occurs within 60 seconds of the last action, the telephone automatically exits from the selection display mode. The following events occur:

- The LED light for the feature button goes out. Or on telephones with feature icons, the feature icon changes to the off state.
- The displayed message is cleared.
- The display of any active call is restored.

When the telephone is in the selection display mode, the timer is reset if a user presses:

- the feature button
- the **Next** button
- any digit key

- Silent ringing

While a telephone is in the message posting mode, the user hears a burst of ringing, and then the telephone rings silently. Silent ringing removes the ringing disturbance while the user can still answer the incoming call.

Interactions for Posted Messages

This section provides information about how the Posted Messages feature interacts with other features in your system. Use this information to ensure that you receive the maximum benefits of Posted Messages in any feature configuration.

- Display mode interactions

The system cancels the selection display mode if one of the following telephone display modes is activated.

- Normal

The normal mode displays call related information for the active call appearance.

Depending on the type of call, this information can include the name and the number of the calling party or the called party.

The elapsed time feature can be invoked anytime that the display is in Normal mode. The elapsed time feature displays elapsed time in hours, minutes, and seconds.

- Inspect

The inspect mode displays 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

The stored number mode displays one of the following numbers:

- The last number that the user dialed
- The number that is stored in an abbreviated dialing button
- A number that is stored in an abbreviated dialing list
- A number that is assigned to a button that is administered by the Facility Busy Indication feature

- Date and time

The date and time mode displays the current date and the time of day.

- Integrated directory

The integrated directory mode turns off the touchtone signals. The integrated directory mode then allows a user to use the touchtone buttons to enter the name of a system user. After the user enters a name, the display shows the name and the extension.

Integrated directory mode can also use the Call-Disp button. The Call-Disp button automatically returns the call that the currently displayed message requested, or by the currently displayed name and extension.

— Message retrieval

The message retrieval mode retrieves messages for telephone users. If no messages are stored, the display shows NO MESSAGES. A user can retrieve messages even if the user is active on a call.

Message retrieval mode can use three additional buttons:

- The Next Message button retrieves the next message. When the telephone is in retrieval mode, the telephone displays END OF FILE, PUSH NEXT TO REPEAT.
- The Delete button deletes the message that is currently displayed.
- The Call-Disp button automatically returns the call that the currently displayed message requested, or by the currently displayed name and extension.

— Coverage message retrieval

The coverage message retrieval mode retrieves messages for users who have telephones without a display. You must administer retrieval permission for a user to retrieve messages from another user.

The user does not have to lift the handset to retrieve messages. The user can retrieve messages even if the user is active on a call. Coverage message retrieval mode can use three additional buttons:

- The Next Message button retrieves the next message. When the telephone is in retrieval mode, the telephone displays END OF FILE, PUSH NEXT TO REPEAT.
- The Delete button deletes the message that is currently displayed.
- The Call-Disp button automatically returns the call that the currently displayed message requested, or by the currently displayed name and extension.

- Diverted Calls

If a user diverts a call to a telephone that is in message posting mode, the calling party does not receive the posted message. For example,

- User A posts a message.
- User B calls user C.
- The system diverts the call through the Call Coverage or Call Forwarding feature to user A.
- User B does not receive the posted message from user A.

- Group Extensions, Hunt Groups, Terminating Extension Groups (TEG), intercom groups

The Posted Messages feature does not apply to calls generated from dialing a group extension. The group member remains available to receive calls, and the calling party does not see the posted message.

The Posted Messages feature only applies when dialing the group member's own extension.

- No Hold Conference

No Hold Conference allows a user to automatically conference another party while continuing the conversation on the existing call. The new party is automatically added to the existing call upon answer.

If the No Hold Conference feature is in process, the user cannot use the Posted Messages feature. If the telephone is in selection display mode, the user cannot use the No Hold Conference feature.

Posted Messages

Interactions for Posted Messages

- Personal Station Access/Terminal Translation Initialization (PSA/TTI)

After PSA/TTI dissociation, the extension of the telephone becomes an X-port extension. If the Posted Messages feature was previously activated for the extension, the Posted Messages feature is deactivated.

The permanent display for the Posted Messages feature has precedence over the permanent displays for the PSA enhancements feature:

- if the Posted Messages feature and the Personal Station Access feature are both active
- when the telephone is PSA-associated

- Transfer

Under normal conditions, a posted message only appears to a calling party. A posted message does not appear to other parties to whom the call might later be transferred.

A posted message appears to the telephones of all transferees if both these conditions apply:

- if a user activates a posted message while the telephone is ringing
- after the transfer operation is completed

Priority Calling

Use the Priority Calling feature provide a special type of call alerting between internal telephone users, including the attendant. The called party hears a distinctive ringing when the calling party uses Priority Calling.

Detailed description of Priority Calling

This section provides a detailed description of the Priority Calling feature.

You enable Priority Calling for your system. The Class of Service (COS) that you assign to each user determines whether a user can use Priority Calling.

The following types of calls are always priority calls:

- Call coverage consult
- Automatic callback
- Ringback queuing
- Attendant intrusion
- Security violation notification

The system assigns a three-burst ringing-pattern as the default for a Priority Calling call.

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 call waiting ringback tone for a multiappearance telephone if no call appearances are idle. Instead, a caller with a multiappearance telephone hears busy tone. The system generates the call waiting ringback tone if the telephone has an idle call appearance, including the call appearance that is reserved for call origination.

Hardware requirements for Priority Calling

The Priority Calling feature requires the following hardware:

- None

Administering Priority Calling

The following steps are part of the administration process for the Priority Calling feature:

- [Assigning a priority feature button to an attendant console](#)
- [Assigning a priority feature button to a telephone](#)

This section describes:

- Any prerequisites for administering the Priority Calling feature
- The screens that you use to administer the Priority Calling feature
- Complete administration procedures for the Priority Calling feature

Prerequisites for administering Priority Calling

You must complete the following actions before you can administer the Priority Calling feature:

- Administer the Feature-Related System Parameters screen to support the Priority Calling feature on your system.
- Create a Class of Service (COS) that allows your users to use Priority Calling.
- Ensure that a feature access code (FAC) for Priority Calling is available on your system.

To administer the **Feature-Related System Parameters** screen to support the Priority Calling feature on your system:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the **Feature-Related System Parameters** screen ([Figure 250, Feature-Related System Parameters screen](#), on page 928).

Figure 250: Feature-Related System Parameters screen

display system-parameters features	Page 5 of 14
FEATURE-RELATED SYSTEM PARAMETERS	
Public Network Trunks on Conference Call: 5	Auto Start? n
Conference Parties with Public Network Trunks: 6	Auto Hold? y
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): 0	Mode Code Interface? y
Reset Shift Timer (seconds): 0	
Station Call Transfer Recall Timer (seconds): 0	
DID Busy Treatment: tone	
Invalid Number Dialed Intercept Treatment: tone	
Allow AAR/ARS Access from DID/DIOD? n	
Allow ANI Restriction on AAR/ARS? n	
Use Trunk COR for Outgoing Trunk Disconnect? n	
7405ND Numeric Terminal Display? n	7434ND? y
DISTINCTIVE AUDIBLE ALERTING	
Internal: 1 External: 2 Priority: 3	
Attendant Originated Calls: external	
DTMF Tone Feedback Signal to VRU - Connection:	Disconnection:

- 2 Page through the screens until you see the **DISTINCTIVE AUDIBLE ALERTING** area.

- 3 In the **Priority** field in the **DISTINCTIVE AUDIBLE ALERTING** area, type **priority** next to the number of rings that you want the system to use for a priority call.

For virtual stations, the number of rings applies to the mapped-to physical telephone.

- 4 Press **Enter** to save your changes.

To create a COS that allows your users to use the Priority Calling feature:

For more information about how to create a COS, see the “Class of Service” feature.

To ensure that a FAC for Priority Calling is available on your system:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the **Feature Access Codes (FAC)** screen ([Figure 251, Feature Access Code \(FAC\) screen](#), on page 929).

Figure 251: Feature Access Code (FAC) screen

change feature-access-codes	FEATURE ACCESS CODE (FAC)	Page 3 of 7
PASTE (Display PBX data on Phone) Access Code: Personal Station Access (PSA) Associate Code: *98 Dissociate Code: *99 Per Call CPN Blocking Code Access Code: Per Call CPN Unblocking Code Access Code: Posted Messages Activation: Deactivation: Priority Calling Access Code: *76 Program Access Code: Refresh Terminal Parameters Access Code: Remote Send All Calls Activation: Deactivation: Self Station Display Activation: *26 Send All Calls Activation: Deactivation: Station Firmware Download Access Code: *36 Station Lock Activation: Deactivation: Station Security Code Change Access Code: *47 Station User Admin of FBI Assign: Remove: Station User Button Ring Control Access Code: Terminal Dial-Up Test Access Code:		

- 2 Click **Next** until you see the Priority Calling Access Code field.
- 3 Type the FAC that you want to use for Priority Calling. In this example, the FAC is *76.
- 4 Press **Enter** to save your changes.

For more information, see the “Feature Access Code” feature.

Screens for administering Priority Calling

Screen name	Purpose	Fields
<i>Attendant Console</i>	Assign a priority feature button to an attendant console.	Feature Button Assignments - priority
<i>Class of Service</i>	Define a COS that allows Priority Calling.	Priority Calling
<i>Feature Access Code (FAC)</i>	Define a FAC for Priority Calling.	Priority Calling Access Code

Screen name	Purpose	Fields
<i>Feature-Related System Parameters</i>	Assign the number of rings for a priority call.	Distinctive Audible Alerting - Priority
<i>Station</i> (multiappearance)	Assign a feature button for Priority Calling to a multiappearance telephone extension.	Button/Feature Button Assignments - priority

Assigning a priority feature button to an attendant console

To assign a priority feature button to an attendant console:

- 1 Type **change attendant *n***, where *n* is the number of the attendant console to which you want to assign a priority feature button. Press **Enter**.

The system displays the *Attendant Console* screen ([Figure 252, Attendant Console screen](#), on page 930).

Figure 252: Attendant Console screen

change attendant 1	ATTENDANT CONSOLE	Page 3 of 4
FEATURE BUTTON ASSIGNMENTS		
1: split	13:	
2:	14:	
3:	15:	
4:	16:	
5:	17:	
6: hold	18:	
7:	19: forced-rel	
8:	20:	
9:	21:	
10:	22:	
11:	23: night-serv	
12: priority	24: pos-busy	

- 2 Page through the screens until you see the FEATURE BUTTON ASSIGNMENTS area.
- 3 Type **priority** next to the feature button that you want the attendant to use for Priority Calling.
- 4 Press **Enter** to save your changes.

Assigning a priority feature button to a telephone

To assign a priority feature button to a telephone:

- 1 Type **change station *n***, where *n* is the extension of a multiappearance telephone to which you want to assign a priority feature button. Press **Enter**.

The system displays the **Station** screen ([Figure 253, Station screen](#), on page 931).

Figure 253: Station screen

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	7:	
2: call-appr	8:	
3: call-appr	9:	
4: priority	10: dn-dst	
5:	11: goto-cover	
6:	12: send-calls Ext:	

- 2 Page through the screens until you see the **BUTTON ASSIGNMENTS** area.
- 3 Type **priority** next to the button assignment that you want the user to use for Priority Calling.
- 4 Press **Enter** to save your changes.

End-user procedures for Priority Calling

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. A user can activate Priority Calling before or after the user places a call. This section includes the following end-user procedures for Priority Calling:

- To activate Priority Calling before the user places a call, a user:
 - 1 Dials the priority calling Feature Access Code (FAC) and an extension
 - 2 Presses a priority feature button and an extension
- To activate Priority Calling after the call starts to ring at the destination, a user:
 - Presses the priority feature button

Reports for Priority Calling

The following reports provide information about the Priority Calling feature:

- None

Considerations for Priority Calling

This section provides information about how the Priority Calling feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Priority Calling under all conditions. The following considerations apply to Priority Calling:

- None

Interactions for Priority Calling

This section provides information about how the Priority Calling feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Priority Calling in any feature configuration.

- Abbreviated Dialing
To place a priority call to an extension on an abbreviated dial list, the user must use a button to which both the FAC for Priority Calling and Abbreviated Dialing are assigned.
- Call Coverage
The system redirects a call to coverage if the user activates Go to Cover for the call. When the call goes to coverage, the call remains a priority call. The covering user hears the priority call ringing pattern.
- Call Forwarding All Calls
The system forwards priority calls, except callback calls. When the system forwards a call, the 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 call waits on an active single-line telephone, even if Call Waiting is not assigned to the telephone. The user with an active, single-line telephone who receives the call, hears the distinctive 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 (DCS)

With 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 the user places the call.

- Last Number Dialed

A user must use the Last Number Dialed button to place a priority call to the last number dialed. The Last Number Dialed FAC is not valid after a user activates Priority Calling.

You can administer single-line telephones, for example, 2500-series telephones, so that the system does not provide distinctive ringing. If you administer single-line telephones in this way, the system provides one-burst ringing for priority calls.

Privacy

Use the Privacy feature to protect your call from interruptions.

Privacy supports the following capabilities:

- **Data Privacy for voice or data calls**
Prevents voice or data calls from being disturbed by any overriding or ringing features
- **Data Restriction for voice or data calls**
Prevents voice or data calls from being disturbed by an overriding or ringing, features or system-generated tones.
- **Privacy-Automatic Exclusion**
Automatically prevents other multiappearance users from bridging onto a call
- **Manual Exclusion**
Prevents other multiappearance users from bridging onto a call when the recipient of the call presses the exclusion button

Detailed description of Privacy

This section provides a detailed description of the Privacy feature.

Data Privacy

The Data Privacy capability prevents analog data calls from being disturbed by any overriding or ringing features. You administer Data Privacy for each user.

Data Restriction

The Data Restriction capability prevents voice or data calls from being disturbed by any overriding or ringing feature or by system-generated tones. You can administer Data Restriction for either a user or a trunk group. When you administer Data Restriction for a voice telephone, data terminal, or a trunk group, the capability is active for all calls to or from those facilities.

Privacy - Automatic Exclusion

Privacy - Automatic Exclusion allows a user of a multiappearance telephone to keep others with appearances of the same extension from bridging onto an existing call. Automatic exclusion is available as soon as a user answers a call. To turn off automatic exclusion, the user presses the exclusion button.

Privacy - Manual Exclusion

Privacy - Manual Exclusion allows a user of a multiappearance telephone to keep others with appearances of the same extension from bridging onto an existing call. To use manual exclusion, the user presses the exclusion button, either before the user places the call, or when the user is active on the call. If the user presses the exclusion button while others are bridged onto the call, the system drops the other users. To turn off manual exclusion, the user presses the exclusion button.

Hardware requirements for Privacy

The Privacy feature requires the following hardware:

- None

Administering Privacy

The following steps are part of the administration process for the Privacy feature:

- [Administering Privacy for a user](#)
- [Activating Data Restriction for a trunk group](#)

This section describes:

- Any prerequisites for administering the Privacy feature
- The screens that you use to administer the Privacy feature
- Complete administration procedures for the Privacy feature

Prerequisites for administering Privacy

You must complete the following actions before you can administer the Privacy feature:

- Ensure that the Automatic Exclusion field on the *Feature-Related System Parameters* screen is set to y.

To view the *Feature-Related System Parameters* screen, type **change system-parameters features**. Press **Enter**.

- Ensure that the Data Privacy Access Code field on the *Feature Access Codes (FAC)* screen is set to y.

To view the *Feature Access Codes (FAC)* screen, type **change feature-access-codes**. Press **Enter**.

- Ensure that the Data Privacy and the Automatic Exclusion fields on the *Class of Service* screen are set to y.

To view the *Class of Service* screen, type **change cos**. Press **Enter**.

Screens for administering Privacy

Screen name	Purpose	Fields
<i>Feature Access Code</i>	Set the access code for Data Privacy.	Data Privacy Access Code
<i>Feature-Related System Parameters</i>	Activate Privacy-Automatic Exclusion.	Automatic Exclusion by COS.
<i>Class of Service</i>	Enable data privacy for a Class of Service (COS).	Data Privacy
<i>Station</i>	Administer COS.	COS
	Administer feature buttons.	BUTTON ASSIGNMENTS: • Exclusion
	Activate Data Restriction.	Data Restriction?
<i>Trunk Group - all</i>	Activate Data Restriction.	Data Restriction?

Administering Privacy for a user

To administer Privacy for a user:

- 1 Type **change station *n***, where *n* is the extension of the user for whom you want to administer Privacy. Press Enter.

The system displays the **Station** screen ([Figure 254, Station screen](#), on page 937), ([Figure 255, Station screen](#), on page 938), and ([Figure 256, Station screen](#), on page 938).

Figure 254: Station screen

change station		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?	
Model:		
Survivable GK Node Name:	Media Complex Ext:	
	IP Softphone? y	
	Remote Office Phone? y	

Figure 255: Station screen

change station		Page 2 of X
STATION		
FEATURE OPTIONS		
LWC Reception? 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	Restrict Last Appearance? y	
Bridged Call Alerting? n		
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:	Select Last Used Appearance? n	
AUDIX Name:	Coverage After Forwarding? _	
Recall Rotary Digit? n	Multimedia Early Answer? n	
Remote Softphone Emergency Calls: as-on-local	Direct IP-IP Audio Connections? n	
Emergency Location Ext: 75001	IP Audio Hairpinning? n	

Figure 256: Station screen

change station Page 3 of 4		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	7: disp-chrg	
2: call-appr	8:	
3: call-appr	9:	
4:	10:	
5:	11: exclusion	
6: headset	12: release	

- 2 In the COS field, type the number of the COS that supports Data Privacy.
- 3 Page through the screens until you see the Data Restrictions? field.
- 4 In the Data Restrictions? field, type y.
If the Auto Answer field is set to **all** or **acd**, you must not set the Data Restriction? field to y.

- 5 Page through the screens until you see the **BUTTON ASSIGNMENTS** area.
- 6 In the **BUTTON ASSIGNMENTS** area, type **exclusion** next to the feature button number that you want the user to use activate privacy-manual exclusion, and to deactivate both privacy-manual exclusion and privacy-automatic exclusion.
- 7 Press **Enter** to save your changes.

Activating Data Restriction for a trunk group

To activate Data Restriction for a trunk group:

- 1 Type **change trunk-group n**, where **n** is the number of the trunk group for which you want to activate Data Restriction.

The system displays the **Trunk-Group** screen ([Table 257, Trunk Group screen](#), on page 939).

Figure 257: Trunk Group screen

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: ____	UII IE Treatment: ____	
	Maximum Size of UII IE Contents: ____	
	Replace Restricted Numbers? _	
	Replace Unavailable Numbers? _	
	Send Connected Number: _	
Send UII IE? _		
Send UCID? _	BRS Reply-best DISC Cause Value: ____	
	Ds1 Echo Cancellation? _	
	US NI Delayed Calling Name Update? _	
	Network (Japan) Needs Connect Before Disconnect? _	
Time (sec) to Drop Call on No Answer: _		
Outgoing ANI: _		
R2 MFC Signaling: _		



CAUTION:

Do not change fields on this page of the **Trunk Group** screen without assistance from Avaya or your network service provider.

- 2 In the **Data Restriction?** field, type **y**.
- 3 Press **Enter** to save your changes.

End-user procedures for Privacy

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Privacy:

- Using Privacy

To use privacy:

- Dial the feature access code (FAC) for Data Privacy at the beginning of the call to activate Data Privacy.
- Press the exclusion button before or during a call to activate Privacy-Manual Exclusion.
- Press the exclusion button to deactivate Privacy-Manual Exclusion or Privacy-Automatic Exclusion.

Reports for Privacy

The following reports provide information about the Privacy feature:

- None

Considerations for Privacy

This section provides information about how the Privacy feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Data Privacy under all conditions. The following considerations apply to Privacy:

- 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 that originate from attendant consoles.
- For virtual extensions, assign the Data Privacy Class of Service to the mapped-to physical extension.
- Do not administer Data Restrictions for an attendant console.

Interactions for Privacy

This section provides information about how the Privacy feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Privacy in any feature configuration.

- **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 that is 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.

When Privacy-Manual Exclusion is active, the system does not allow other users to bridge onto the active call.
- **Busy Verification**
Busy Verification cannot be active when Data Privacy is active.
- **Call Coverage**
When the principal user bridges onto a call that has gone to coverage, and the call is answered at the coverage point, the system does not drop the principal user from the call when Privacy-Manual Exclusion is activated.
- **Call Pickup**
The system does not drop the called party from the call in the following example:
 - A call is made to user A.
 - User B uses Call Pickup to answer the call.
 - User A bridges onto the call by going off-hook on its call appearance.
 - User B activates Privacy-Manual Exclusion.
- **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. This action prevents 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.
- **Whisper Paging**
If you administer Data Restriction for a telephone, a data terminal, or a trunk group, the system denies Whisper Paging.

Public Network Call Priority

Use the Public Network Call Priority feature to provide call retention, forced disconnect, intrusion, mode-of-release control, and re-ring to servers on public networks.

Detailed description of Public Network Call Priority

This section provides a detailed description of the Public Network Call Priority feature.

Use the Public Network Call Priority feature to provide call retention, forced disconnect, intrusion, mode-of-release control, and re-ring to servers on public networks. Different countries refer to these functions by different names. Not all functions are available in all countries.

China

Forced disconnect

With forced disconnect network operator can disconnect a called party from a local call, and connect the called party to an incoming toll call. Parties on the local call hear a warning tone before the network operator disconnects the call. forced disconnect is allowed only for callers on local single-station calls. The system does not allow forced disconnect for the following type of calls:

- Conference
- Transferred
- Forwarded
- To group users
- Tandem

Mode-of-release control

Mode-of-release control inhibits release of a trunk circuit when a caller goes on-hook, based on call type and direction. Instead of releasing the trunk circuit, the system keeps the circuit active, and reconnects the call if the caller goes off-hook again. Mode-of-release control applies to the following types of incoming or outgoing calls:

- Toll
- Local
- Service

Public Network Call Priority provides three types of mode-of-release control.

- **Calling-party control**

When calling-party control is active, the trunk is not released until the caller goes on-hook. For example, if the:

- Caller goes on-hook, the trunk is released immediately. The called party receives busy tone.
- Called party goes on-hook, the trunk is not released until the caller goes on-hook or the re-answer timer for outgoing calls expires. The called party can re-answer the call, and talk to the calling party.

Re-ring occurs for incoming calls to the system with calling-party control. When the called party goes on-hook, the trunk is not released, and the central office (CO) operator can re-ring the called party and reconnect the call.

- Re-answer timer is activated and expired, the trunk is released on outgoing calls with calling-party control.

- **Called-party control**

When called-party control is active, the trunk is not released until the called party goes on-hook.

- If the called party goes on-hook, the trunk is released immediately, and the caller receives busy tone.
- If the caller goes on-hook, the trunk is not released until the called party goes on-hook. The caller can go off-hook again to reconnect. No timer is involved with called-party control

- **First-party control**

When first-party control is active, the trunk is released immediately regardless of whether the caller or the called party goes on-hook first. The party that is still connected receives busy tone. The default or normal mode-of-release control for the system is first-party control.

Russia

Intrusion

Intrusion allows a network operator to break into a local call and announce an incoming toll call. Intrusion is allowed on local single-line and multiline telephone calls. The system does not allow intrusion for the following types of calls:

- Conference
- On hold
- Toll

Re-ring

Re-ring occurs when a call is interrupted by an operator-assisted incoming call, and the call is kept on hold so that the call can be reconnected to a telephone. When the called party goes on-hook, the network toll operator can re-ring the called party, and reconnect the call.

Spain

Call retention

When a caller makes an emergency call and then hangs up, the call is put on hold. The system does not disconnect the call. When the caller goes back off-hook, the telephone reconnects to the emergency call. Call retention works on both analog and digital telephones.

Re-ring

Re-ring occurs when a call is interrupted by an operator-assisted incoming call, and the call is kept on hold so that the call can be reconnected to a telephone. When the called party goes on-hook, the network toll operator can re-ring the called party, and reconnect the call.

Hardware requirements for Public Network Call Priority

The Public Network Call Priority feature requires the following hardware:

- None

Administering Public Network Call Priority

This section describes the screens that you use to administer the Public Network Call Priority feature

Screens for administering Public Network Call Priority

Screens for China

Screen name	Purpose	Fields
<i>Multifrequency-Signaling-Related System-Parameters</i>	Specify the type of tone that is received from a Chinese central office (CO).	Incoming Forward Signal Types for group I and group II
	Specify the type of tone that is sent to a Chinese CO.	Incoming Backward signal Types for group A and group B
<i>Trunk Group</i>	Specify country code 18 for China.	Country
	Specify the outgoing dial type of mf for China.	Outgoing Dial Type
	Specify the incoming dial type of mf for China	Incoming Dial Type

Screens for Russia

Screen name	Purpose	Fields
<i>Trunk Group (DID)</i>	Specify country code 15 for Russia.	Country
	Specify the protocol type Intol for Russia.	Protocol Type
<i>Trunk Group (DIOD)</i>	Specify country code 15 for Russia.	Country
	Specify the protocol type Intol for Russia.	Protocol Type

Screens for Spain

Screen name	Purpose	Fields
<i>Trunk Group</i>	Specify country code 11 for Spain.	Country

Reports for Public Network Call Priority

The following reports provide information about the Public Network Call Priority feature:

- None

Considerations for Public Network Call Priority

This section provides information about how the Public Network Call Priority feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Public Network Call Priority under all conditions. The following considerations apply to Public Network Call Priority:

- None

Interactions for Public Network Call Priority

This section provides information about how the Public Network Call Priority feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Public Network Call Priority in any feature configuration.

China

Forced disconnect

- Conference
If the network toll call terminates at a telephone that is involved in a conference, the network does not send the forced disconnect signal.
- Call Forwarding
The system does forward the forced disconnect signal for calls that are forwarded on-premises, on the network, or off the network.
- Group Users
The network does not send the forced disconnect signal, if a network toll call terminates to a group user.
- Non-Station Users
The network does not send the forced disconnect signal, if a network toll call terminates to a non-station user.
- Tandem Trunks
The system does not tandem a forced disconnect signal.
- Transfer
The network does not send the forced disconnect signal, if a network toll call is transferred.

Mode-of-release control

- Conference
A call that is involved in a conference is changed to first-party control as the mode-of-release control.
- Forward
A forwarded call on-premises, on the network, or off the network is changed to first-party control as the mode-of-release control.
- Group users - hunt, trunk, terminating extension group (TEG), Intuity AUDIX, Vector Directory Number (VDN)
Calls that terminate to group users are changed to first-party control as the mode-of-release control.
- Nonstation users - personal attendant, data-module, announcement, voice synthesis
Calls that terminate to nonstation users are changed to first-party control as the mode-of-release control.
- Tandem Trunks
The system terminates tandem calls, but the mode-of-release control is changed to first-party control.
- Transfer
A transferred call is changed to first-party control as the mode-of-release control.

Re-ring

- Conference
A call that is involved in a conference is changed to first-party control as its mode-of-release control. first-party control calls do not re-ring.
- Call Forwarding
The system does not forward re-ring signals for calls forwarded on-premises, on the network, or off the network.
- Group users - hunt, trunk, TEG, AUDIX, and VDN
The system ignores Re-ring signals that are sent to group users.
- Nonstation user - personal attendant, data-module, announcement, voice synthesis
The system ignores Re-ring signals that are sent to nonstation users.
- Tandem trunks
The system does not tandem re-ring signals.
- Transfer
A transferred call is changed to first-party control as its mode-of-release control. First-party control calls do not re-ring.

Russia

- Announcements
The system does not allow intrusion and re-ring for an announcement port.
- Abbreviated Ringing and Delayed Ringing
Abbreviated Ringing and Delayed Ringing characteristics that you assign do not apply to re-ring. Re-ring has its own priority ringing.
- Administered Connections
Intrusion and re-ring do not apply to Administered Connections.
- Attendant Console
Intrusion and re-ring do not apply to attendant consoles, or to any call that involves an attendant console.
- Attendant Serial Call
The system ignores Intrusion and re-ring for an attendant serial call.
- Automatic Callback
Re-ring takes precedence over automatic callback on busy or no-answer calls.
- Busy Verification and Attendant Intrusion
While Intrusion or re-ring occurs, Busy Verification and Attendant Intrusion are denied. While Busy Verification or Attendant Intrusion occurs, Intrusion and re-ring are denied.
- Call Coverage
Re-ring overrides Call Coverage. However, if a station is busy and a coverage destination is free, an incoming toll call rings at the coverage destination instead of intruding on the busy call.

- **Call Forwarding**
Intrusion can be used with Call Forwarding. If a station is busy, an incoming toll call is forwarded instead of intruding on the busy call. Re-ring overrides all administered redirection.
- **Call Waiting**
If Call Waiting is active, calls are not intruded upon. Call Waiting takes precedence.
- **Conference**
The system does not allow intrusion for a call that is involved in a conference.
- **Data Calls**
The system does not allow intrusion for telephones that have Data Privacy, Data Restriction, or Data Protection active.
- **Distinctive Ringing**
Ringing characteristics that you assign do not apply to re-ring. Re-ring has its own priority ringing.
- **Do Not Disturb**
You cannot use intrusion and Do Not Disturb at the same time.
- **Emergency Access to the Attendant**
The system does not allow intrusion for an emergency call.
- **Hunt Group and Automatic Call Distribution**
If a hunt group queue is not busy, the system places incoming toll calls in the queue. Busy calls are not intruded upon.
- **Intercom - Automatic and Dial**
The system does not allow intrusion on any Intercom calls.
- **Malicious Call Trace (MCT)**
The system does not allow intrusion on a station that has MCT active.
- **Personal Station Access (PSA)**
The system does not allow Intrusion when PSA is in use.
- **Pull Transfer**
The system does not allow Intrusion when Pull Transfer is in use.
- **Class of Restriction (COR)**
The system does not allow Intrusion, regardless of the COR.
- **Ringback Queuing**
Intrusion is allowed with Ringback Queuing.
- **Station Hunting**
The system allows Intrusion when Station Hunting is used.
- **Tandem Trunks**
The system does not allow Intrusion over trunk groups used as tandem trunks.

Recorded Telephone Dictation Access

Use the Recorded Telephone Dictation Access feature to access dictation equipment.

Detailed description of Recorded Telephone Dictation Access

This section provides a detailed description of the Recorded Telephone Dictation Access feature.

Use the Recorded Telephone Dictation Access feature to access dictation equipment. Users can access the feature from any on-site or off-site telephone, and can use incoming tie trunks to access the feature.

A user enters a Feature Access Code (FAC) or an extension to access the feature. A user can enter commands by key or by voice to control the start and stop functions. A user can enter commands by key to control other functions, for example, initial activation.

Recorded Telephone Dictation Access cannot be used with Automatic Route Selection (ARS) or Conference.

For more information, see the following features:

- “Audible Message Waiting”
- “Announcements”
- “Voice Message Retrieval”

Hardware requirements for Recorded Telephone Dictation Access

The Recorded Telephone Dictation Access feature requires the following hardware:

- None

Reports for Recorded Telephone Dictation Access

The following reports provide information about the Recorded Telephone Dictation Access feature:

- None

Considerations for Recorded Telephone Dictation Access

This section provides information about how the Recorded Telephone Dictation Access feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Recorded Telephone Dictation Access under all conditions. The following considerations apply to Recorded Telephone Dictation Access:

- None

Interactions for Recorded Telephone Dictation Access

This section provides information about how the Recorded Telephone Dictation Access feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits o

- Automatic Route Selection (ARS)
You cannot use ARS to access Recorded Telephone Dictation Access.
- Conference
You cannot use Conference and Recorded Telephone Dictation Access at the same time.

Remote Access

Use the Remote Access feature to access and use the system from the public network.

Detailed description of Remote Access

This section provides a detailed description of the Remote Access feature.



SECURITY ALERT:

Avaya has designed the Remote Access feature incorporated in this product, when properly administered by the customer, to enable the customer to minimize the ability of unauthorized people to gain access to the network. It is the responsibility of the customer to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes, and distribute the access codes only to people whom you advise of the sensitive nature of the access information. Instruct each authorized user to use access codes properly.

In rare instances, unauthorized individuals use the Remote Access feature to make connections to a telecommunications network. 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.

The Remote Access caller must access your system from the public network and use a touch tone telephone or equivalent equipment. When a user uses Remote Access, the system does not have access to the calling number, because the calling number is outside the system. Thus, some features and capabilities, such as Ringback Queuing and Automatic Callback, cannot be used on a Remote Access call. Also, the user cannot use any feature that requires recall dial tone, such as, Hold and Transfer, from the remote location.

Remote Access provides users with access to the system and system features from the public network. Users can use Remote Access to make business calls from home or use the Recorded Telephone Dictation Access to dictate a letter. An authorized user can also access system features from any on-site extension.

With Remote Access, you can dial into the system over direct inward dialing (DID), central office (CO), foreign exchange (FX), or 800 service trunks. When a call comes in on a trunk group that is dedicated to Remote Access, the system routes the call to the Remote Access extension that you assigned. If DID is provided, and the Remote Access extension is within the range of numbers that can be accessed by DID, the system uses DID for Remote Access.

You can administer your system so that a user must enter a barrier code, an authorization code, or both to use Remote Access.

Use barrier codes to secure and define calling privileges through the Class of Restriction (COR) that you administer to users and trunk groups. You can administer as many as 10 barrier codes. Each barrier code has a different COR and Class of Service (COS). Barrier codes can be from 4 to 7 digits, *but all barrier codes that you define must be the same length.*

Night Service

You can administer your system to provide attendant-assisted calling during the day, and then Remote Access when the system is in Night Service.

Security

To ensure system security, you can permanently disable the Remote Access feature if you do not intend to use the feature. If you permanently disable Remote Access, you must contact your Avaya representative to activate this feature again.



CAUTION:

Your attempt to disable the Remote Access feature is lost if the server that runs Avaya Communication Manager is rebooted without saving translations. Therefore, you must run a “save translation” command after you permanently disable the Remote Access feature.

The system provides several means to secure your system when you use the Remote Access feature:

- The **status remote-access** command
- Barrier codes
- Authorization codes
- Alternate Facility Restrictions Levels (AFRLs)
- COR
- Logoff Notification

Status remote-access

You can check the status of the Remote Access feature and the barrier codes. The **status remote-access** command displays information that can help you determine when and why the system denied remote access to a user, or why the system blocked a barrier code. When you type the **status remote-access** command, the system displays the:

- Remote Access status:
 - Not administered
 - Enabled
 - Disabled
 - Disabled following detection of a security violation
- Date and time that Remote Access was last modified
- Barrier code information:
 - Date that the code was administered, reactivated, or modified
 - Expiration date
 - Number of calls that can be placed with the code
 - Number of calls that were placed with the code
 - Active or expired status
 - The Date and the reason that a code expired

For a detailed description of the **status remote-access** command and display, see the *BCS Products Security Handbook*.

Barrier codes

Remote Access has inherent risks, such as large-scale unauthorized long distance use of your telecommunications facilities. To increase the security of your system, use a 7-digit barrier code, and administer expiration dates and access limits for each of the 10 barrier codes that are available to you. If your system has more than 10 Remote Access users, the users must share barrier codes. A barrier code automatically expires if the expiration date or the number of accesses exceeds the limits that you set. You can administer the system to limit:

- The length of time that an access code remains valid
- The number of times that an access code can be used
- Both the length of time that an access code remains valid and the number of times that an access code can be used

When you no longer need a barrier code, remove the code from the system.

If you administer barrier codes, a special answer-back tone causes a calling modem to leave dial mode. Sometimes a modem dialer is used to gain access with Remote Access. When a dialer of a modem is used to gain access with Remote Access, the special answer-back tone cancels echo suppressors on the network. The cancellation of echo suppressors in the network prevents dual-tone multifrequency (DTMF) tones from breaking dial tone from Avaya Communication Manager.

Use the **status remote-access** command to view the status of a Remote Access barrier code.

Call Detail Recording (CDR) does not track the use of barrier codes.

Authorization codes

You can administer authorization codes to manage access to your system. You can then use Call Detail Recording to track the use of authorization codes. Use the following guidelines to manage the use of authorization codes.

- Assign authorization codes that:
 - Are random, nonconsecutive, and unpredictable
 - Are the maximum code length that the system allows
 - Are unique to each person who uses an authorization code
 - Have the minimum level of calling permissions that a user requires
- Change codes frequently, at least quarterly.
- Delete codes when the codes are no longer needed.
- Delete codes when a user leaves the company, or changes job assignments.
- Use CDR reports to monitor and analyze the use of the codes.

Alternate Facility Restriction Levels (AFRLs)

Consider the use of AFRLs instead of Facility Restriction Levels (FRLs) after normal business hours to restrict where calls can be made over your facilities. Do not restrict callers from summoning emergency services after normal business hours.

Class of Restriction

The COR of an authorization code supersedes the COR of a barrier code.

- Time of Day Routing is controlled by the time-of-day entries in the COR or by the partition.
- Toll Restriction and Analysis is controlled by COR.
- Trunk Access Code (TAC) interacts with toll restriction. You can translate Communication Manager so that users can use ARS to make toll calls, without the need for a TAC.

For additional steps to secure your system, and to obtain security information on a regular basis, see the *Avaya Toll Fraud and Security Handbook*.

Logoff Notification

Use Logoff Notification when you enable Remote Access for your system, but your users are not actively using Remote Access. Logoff Notification notifies you when you log off the system that Remote Access is enabled.

Logoff Notification alerts you to an unauthorized activation of the Remote Access feature. Logoff Notification is administered by login ID.

Hardware requirements for Remote Access

The Remote Access feature requires the following hardware:

- None

Administering Remote Access

The following steps are part of the administration process for the Remote Access feature:

- [Administering the Remote Access screen](#)
- [Assigning authorization codes](#)
- [Administering Remote Access for Night Service](#)

This section describes:

- Any prerequisites for administering the Remote Access feature
- The screens that you use to administer the Remote Access feature
- Complete administration procedures for the Remote Access feature

Prerequisites for administering Remote Access

You must complete the following actions before you can administer the Remote Access feature:

- None

Screens for administering Remote Access

Screen name	Purpose	Fields
Authorization Code - COR Mapping	Assign pairs of authorization codes and Classes of Restriction (CORs).	• AC • COR
Feature-Related System Parameters	Assign the length of the authorization codes.	Authorization Code Length
Optional Features	Enable authorization codes.	Authorization Codes?
Remote Access	Enable Remote Access.	All
	Disable Remote Access	Permanently Disable?
Trunk Groups • CO • DID • FX • ISDN-BRI • ISDN-PRI • WATS	Assign Remote Access for Night Service.	• Incoming Destination • Night Services

Administering the Remote Access screen

To administer Remote Access, you must complete the following procedures:

- Enable Remote Access
- Disable Remote Access

Enabling Remote Access

To Enable Remote Access:

- 1** Type **change remote-access**. Press **Enter**.

The system displays the **Remote Access** screen ([Figure 258, Remote Access screen](#), on page 958).

Figure 258: Remote Access screen

change remote-access

REMOTE ACCESS

Remote Access Extension: _____
Barrier Code Length: _____

Authorization Code Required? **y**
Remote Access Dial Tone: **n**

Barrier Code	COR COR	TN TN	COS 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)

- 2 In the Authorization Code Required field, perform one of the following actions:
 - If you want a user to enter an authorization code to use Remote Access, type **y**.
 - If you do not want a user to enter an authorization code when the user uses Remote Access, type **n**.

The **Calls Used** field is a display-only field that shows the number of calls that are placed with the corresponding barrier code. The system increments this field whenever a barrier code is successfully used to access the Remote Access feature.

- 3 In the **Barrier Code** field, perform one of the following actions:
 - Type a barrier code.
The number that you type in the **Barrier Code Length** field determines the number of digits that you type in this field.
 - If the **Barrier Code Length** field is blank, you must type **none** in the first **Barrier Code** field.

You can assign 10 barrier codes to your system. You cannot assign duplicate barrier codes.

- 4 In the **Barrier Code Length** field, type the length of the barrier codes that you want to use in your system. Valid entries are the numbers **4** through **7**. You can also leave this field blank.
- 5 In the **COR** field, type the COR number that is associated with the barrier code. The barrier code defines the call restriction features.
- 6 In the **COS** field, type the COS number that is associated with the barrier code. The barrier code defines access permissions for Call Processing features. Valid entries are the numbers **0** to **15**.

- 7 In the **Disable Following a Security Violation** field, perform one of the following actions:
- If you want the system to disable the Remote Access feature when the system detects a remote access security violation, type **y**.
 - If you do not want the system to disable the Remote Access feature when the system detects a remote access security violation, type **n**.

The system displays the **Disable Following a Security Violation** field when the **SVN Authorization Code Violation Notification Enabled** field on the **Security-Related System Parameters** screen is set to **y**

- 8 In the **Expiration Date** field, type the date that you want the barrier code to expire. You must type a date that is greater than the current date. You can also leave the field blank.
- If you assign an expiration date, the system displays a warning message on the **System Copyright** screen 7 days before the expiration date of the barrier code. If you want to extend the expiration date, change the date in this field.

- 9 In the **No. of Calls** field, type the number of times that users can use the barrier code for Remote Access. Valid entries are the numbers **1 to 9999**.

- 10 In the **Permanently Disable?** field, type **n**.

- 11 In the **Remote Access Dial Tone** field, perform one of the following actions:

- If you want the system to provide a Remote Access dial tone prompt, type **y**.
- If you do not want the system to provide a Remote Access dial tone prompt, type **n**.

To maintain system security, Avaya recommends that you set this field to **n**.

The system displays this field only when the **Authorization Code Required** field is set to **y**.

- 12 In the **Remote Access Extension** field, perform one of the following actions:

- If no barrier codes exist, leave the field blank.
- If barrier codes exist, type the Remote Access extension.

The remote access extension is used like a DID extension. Only one DID extension can be assigned as the Remote Access extension. Calls to the Remote Access extension are treated the same as calls on the remote access trunk.

- 13 In the **TN** field, type the Tenant Partition number. Valid entries are the numbers **1 to 100**.

- 14 Press **Enter** to save your changes.

Disabling Remote Access

To disable Remote Access:

- 1 Type **change remote-access**. Press **Enter**.

The system displays the **Remote Access** screen ([Figure 258, Remote Access screen](#), on page 958).

- 2 In the **Permanently Disable?** field, type **n**.

- 3 Press **Enter** to save your changes.

Assigning authorization codes

Prerequisites

You must complete the following actions before you can assign authorization codes:

- Ensure that authorization codes are enabled on the **Optional Features** screen.
- Administer the authorization code parameters on the **Feature-Related System Parameters** screen.

To ensure that authorization codes are enabled on the **Optional Features** screen:

- On the **Optional Features** screen, ensure that the Authorization Codes? field is set to **y**. If the Authorization Codes? field is set to **n**, your system is not enabled for authorization codes. Contact your Avaya representative for assistance before you continue with this procedure

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

For a complete description of the System-Parameters Customer-Options screen, see the *Administrator's Guide for Avaya Communication Manager* for more information.

To administer the authorization code parameters on the **Feature-Related System Parameters** screen:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the **Feature-Related System Parameters** screen.

Figure 259: Feature-Related System Parameters screen

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: <u>tone</u>	
Controlled Termination Restriction (Do Not Disturb): <u>tone</u>	
Controlled Station to Station Restriction: <u>tone</u>	
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 Codes Enabled?** field, perform one of the following actions.
- If you want the users to use authorization codes, type **y**.
 - If you do not want the users to use authorization codes, type **n**.

You can administer this field only if the **Authorization Codes?** field on the **Optional Features** screen is set to **y**.



SECURITY ALERT:

To maintain system security, Avaya recommends that you set the **Authorization Codes Enabled?** field to **y**.

- 3 In the **Authorization Code Length** field, type the length of the authorization codes. Authorization codes must be between 4 and 13 digits long.

The system displays this field only if the **Authorization Codes Enabled?** field is set to **y**.



SECURITY ALERT:

To maintain system security, Avaya recommends that you use the maximum length for the authorization code.

- 4 In the **Authorization Code Cancellation Symbol** field, perform one of the following actions:
- If both the main server and the tandem server are the same type of server, type a pound sign (#).
 - If an Avaya System 85 or a DIMENSION is involved, type the number **1**.

A user dials the authorization code cancellation symbol to cancel the 10-second wait period, during which the user can enter an authorization code

The system displays this field only when the **Authorization Codes Enabled?** field is set to **y**.

- 5 In the **Attendant Time Out Flag** field, perform one of the following actions:
- If you want the system to route a call to the attendant if a user does not dial an authorization code within the 10-second wait period, or if a user dials an invalid authorization code, type **y**.
 - If you do not want the system to generate an intercept tone if a user does not dial an authorization code within the 10-second wait period, or if a user dials an invalid authorization code, type **n**.

The system displays this field only when the **Authorization Codes Enabled?** field is set to **y**.

- 6 In the **Display Authorization Code?** field, perform one of the following actions:
- If you want the system to display the authorization code as the user dials the authorization code, type **y**.
 - If you do not want the system to display the authorization code as the user dials the authorization code, type **n**.

This field applies only to digital communication protocol (DCP) telephones. The field does not apply to ISDN-BRI or hybrid sets.

! SECURITY ALERT:

To enhance the security of your system, Avaya recommends that you set the **Display Authorization Code?** field to **n**.

- 7** Press **Enter** to save your changes.

To assign authorization codes:

- 1** Type **change authorization-code**. Press **Enter**.

The system displays the Authorization Code - COR Mapping screen ([Figure 260, Authorization Code - COR Mapping screen](#), on page 962).

Figure 260: Authorization Code - COR Mapping screen

[illegible]

The **Number of Codes Administered** field is a display-only field. This field contains the number of authorization codes that you administered on the **Authorization Code - COR Mapping** screen. The system limits the number of authorization codes that you can administer. To determine the number of authorization codes that you can administer, type **display capacity**.

- 2** In the **AC** field, type the length of the authorization codes. Authorization codes must be between 4 and 13 digits long.

The number of digits that you type in this field must be the number of digits that you assigned to the Authorization Code Length field on the ***Feature-Related System Parameters*** screen.

- 3 In the COR field, type the Class of Restriction (COR) that the system uses when a user enters the associated authorization code.

- 4** Press **Enter** to save your changes.

Administering Remote Access for Night Service

To administer Remote Access for Night Service:

- 1 Type **change trunk-group *n***, where ***n*** is the number of the trunk group for which you want to administer Remote Access for Night Service. Press **Enter**.

The system displays the **Trunk Group** screen ([Figure 261, Trunk Group screen](#), on page 963).

Figure 261: Trunk Group screen

change trunk-group 3
Page 1 of x

TRUNK GROUP

Group Number: ____	Group Type: tie____	CDR Reports: _
Group Name: _____	COR: ____	TN: ____ TAC: ____
Direction: incoming	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? _	
Incoming Calling Number - Delete: ____	Insert: ____ Format: ____
Bit Rate: ____	Synchronization: ____ Duplex: ____
Disconnect Supervision - In? _ Out? _	
Answer Supervision Timeout: ____	Receive Answer Supervision? _

- 2 In the Incoming Destination field, type **att**.
- The system displays the Incoming Destination field, when the Direction field is set to **incoming** or **two-way**.
- 3 In the Night Service field, type the Remote Access extension.
 - 4 Press **Enter** to save your changes.

End-user procedures for Remote Access

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Remote Access:

- Accessing the attendant

To access the attendant:

- 1** Enter the Remote Access extension.
- 2** Enter the barrier code.
- 3** Enter the attendant access code.

Reports for Remote Access

The following reports provide information about the Remote Access feature:

- None.

Considerations for Remote Access

This section provides information about how the Remote Access feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Remote Access under all conditions. The following considerations apply to Remote Access:

- After the baud of a digital-terminal data module (DTDM) is changed from 9600 to 1200, the DTDM cannot be accessed by Remote Access until an internal call is made to the DTDM.

Interactions for Remote Access

This section provides information about how the Remote Access feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Remote Access in any feature configuration.

- Authorization Codes

When a Remote Access caller dials the assigned Remote Access extension and connects to the system, the system can 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 Class of Restriction (COR) that is assigned to the authorization code supersedes the calling privileges that are assigned to the barrier code.

- Call Detail Recording (CDR)

CDR tracks the use of authorization codes. CDR does not track the use of barrier codes.

- Class of Restriction (COR)

COR restrictions do not block access to the Remote Access feature.

- Night Service

You can specify the Remote Access extension as the Night Service extension on incoming, non-direct inward dialing (DID) trunk groups.

Restriction - Controlled

Use the Restriction - Controlled feature to allow a user with console permission to:

- Activate and deactivate specific restrictions for an individual user or an attendant
- Activate and deactivate specific restrictions for all users or attendants who have a specific Class of Restriction (COR)

Detailed description of Restriction - Controlled

This section provides a detailed description of the Restriction - Controlled feature.

With Restriction - Controlled, a user with console permissions can:

- Activate and deactivate specific restrictions for an individual user or an attendant
- Activate and deactivate specific restrictions for all users or attendants, who have a specific Class of Restriction (COR)

Use Restriction - Controlled to administer the following restrictions:

- Outward

The user cannot place calls to the public network.

- Total

The user cannot place or receive calls, with the following exceptions:

- Calls to a remote-access extension
- Terminating-trunk transmission tests
- Emergency Access to Attendant calls

- Termination

The user cannot receive any calls. The system:

- Routes incoming calls to the attendant
- Redirects calls to the Call Coverage path
- Uses Restriction - Controlled intercept treatment

- Station-to-Station

The user cannot place or receive station-to-station calls.

Hardware requirements for Restriction - Controlled

The Restriction - Controlled feature requires the following hardware:

- None

Administering Restriction - Controlled

This section describes the screens that you use to administer the Restriction - Controlled feature

Screens for administering Restriction - Controlled

Screen name	Purpose	Fields
<i>Feature Access Code (FAC)</i>	Specify the feature access codes (FACs) for Restriction - Controlled.	• User Control Restrict Activation and Deactivation • Group Control Restrict Activation and Deactivation
<i>Feature-Related System Parameters</i>	Specify the type of intercept treatment that the system uses for Restriction - Controlled.	• Controlled Outward Restriction Intercept Treatment • Controlled Termination Restriction (Do Not Disturb) • Controlled Station-to-Station Restriction

End-user procedures for Restriction - Controlled

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Restriction - Controlled:

- Activating Restriction - Controlled

Activating Restriction - Controlled

To activate Restriction - Controlled:

- 1** Enter the feature access code (FAC) that allows you to apply Restriction - Controlled to an extension or an attendant group.
- 2** Enter the number for the type of restriction that you want:
 - 1 for outward
 - 2 for total
 - 3 for termination
 - 4 for station-to-station
- 3** Perform one of the following actions:
 - If you want to apply Restriction Controlled to an individual extension, enter the extension
 - If you want to apply Restriction Controlled to all users and attendant who are assigned a certain Class of Restriction (COR), enter the COR.

Reports for Restriction - Controlled

The following reports provide information about the Restriction - Controlled feature:

- None

Considerations for Restriction - Controlled

This section provides information about how the Restriction - Controlled feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Restriction - Controlled under all conditions. The following considerations apply to Restriction - Controlled:

- All voice terminals with the same COR are affected by a group restriction. When a call is placed, the system checks both the individual and the group restrictions.

Interactions for Restriction - Controlled

This section provides information about how the Restriction - Controlled feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Restriction - Controlled in any feature configuration.

- Call Coverage

The system does not check controlled restrictions for covering users.
- Call Forwarding

The system checks the controlled restrictions for the forwarded-to extension, when Call Forwarding All Calls is active.

Restriction - Controlled

Interactions for Restriction - Controlled

- Class of Restriction (COR)

The system checks the COR when a call is authorized.

- Priority Call

If a user activates priority calling before the user dials another extension, the calling user receives intercept tone. The user receives the intercept tone whether you set **Controlled Station to Station Restriction** field on the ***Feature-Related System Parameters*** form to **y** or **n**.

- Uniform Call Distribution (UCD)

The system does not apply Restriction - Controlled to calls that are dialed through the Uniform Dial Plan (UDP).

Ringling - Abbreviated and Delayed

Use the Ringling - Abbreviated and Delayed feature to assign one of four ring types to each call appearance on a telephone. The ring type that you assign to a call appearance is automatically assigned to each of the bridged call appearances of each call appearance.

Detailed description of Ringling - Abbreviated and Delayed

This section provides a detailed description of the Ringling - Abbreviated and Delayed feature.

The Ringling - Abbreviated and Delayed feature has two categories of ringing:

- Ringing that alerts consistently and does not change:
 - Ringing, in which the lamp flashes and audible ringing occurs
 - Silent ringing, in which the lamp flashes and audible ringing does not occur
- Ringing that transitions from one ringing state to another:
 - Abbreviated ringing, in which ringing continues for the number of cycles that you specify with the automatic abbreviated transition interval or the delayed transition interval, and then changes to silent alerting
 - Delayed ringing, in which visual alerting continues for the number of cycles that you specify with the automatic abbreviated transition interval or the delayed transition interval, and then changes to ringing

When you administer the **Station** screen of a user, you can assign an abbreviated dial button of that user to another user. The user of the **Station** screen that you administer must have a telephone with call appearances that have either abbreviated or delayed ringing. When a call alerts at one of those call appearances, the user presses the button. When the user presses the button, the system forces an immediate transition from ringing to silence, or from silence to ringing.

The Ringling - Abbreviated and Delayed feature is most useful in bridging situations in which some users want to:

- Have a call audibly alert as soon as the call arrives
- Be audibly notified if the call is not answered within a specified number of rings
- Stop the audible alerting if the call is unanswered by the called party, and the user cannot answer the call

You specify the types of ringing on the **Station** screen of each user in your system. You can assign one of the following ring types to each telephone line button.

- Abbreviated Ring

A call rings the telephone until the automatic or the manual automatic abbreviated transition or the delayed transition occurs. After the transition, the call silently alerts at the telephone.

- Delayed Ring
A call silently alerts the telephone until the automatic or the manual abbreviated transition or the delayed transition occurs. After the transition, the call rings at the telephone.
- No Ring
A call silently alerts the telephone and does not transition.
- Ring
A call rings at the telephone, and does not transition.

When a user presses the abbreviated-ring button on the telephone, the system performs an abbreviated transition or a delayed transition for all calls at the extension. Calls to other extensions that alert at the telephone are not affected.

Hardware requirements for Ringling - Abbreviated and Delayed

The Ringling - Abbreviated and Delayed feature requires the following hardware:

- None

Administering Ringling - Abbreviated and Delayed

The following steps are part of the administration process for the Ringling - Abbreviated and Delayed feature:

- [Assigning ring control to a user](#)
- [Assigning an abbreviated ringing button to a user](#)

This section describes:

- Any prerequisites for administering the Ringling - Abbreviated and Delayed feature
- The screens that you use to administer the Ringling - Abbreviated and Delayed feature
- Complete administration procedures for the Ringling - Abbreviated and Delayed feature

Prerequisites for administering Ringling - Abbreviated and Delayed

You must complete the following actions before you can administer the Ringling - Abbreviated and Delayed feature:

- Assign the number of rings before a transition

To assign the number of rings before a transition:

- 1 Type **change system-parameters features**. Press **Enter**.

The system displays the [Figure 262, Feature-Related System Parameters](#), on page 971).

Figure 262: Feature-Related System Parameters

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/SIP 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?

```

- 2 In the Auto Abbreviated/Delayed Transition Interval (rings) field, type the number of rings before the system performs an automatic abbreviated transition or delayed transition for a call. You can type a number from **1** to **16**.
- 3 Press **Enter** to save your changes.

Screens for administering Ringling - Abbreviated and Delayed

Screen name	Purpose	Fields
Feature-Related System Parameters	Assign the number of rings before the system performs an automatic abbreviated transition or a delayed transition for a call.	Auto Abbreviated/Delayed Transition Interval (rings)
Station	Allow a user to select ringing for call appearances.	Per Button Ring Control
	Assign a ringing-Abbreviated and Delayed button to a user.	Feature Buttons

Assigning ring control to a user

To assign ring control to a user:

- 1 Type **change station *n***, where *n* is the number of the extension to which you want to assign ring control for a user. Press **Enter**.

The system displays the [Figure 263, Station screen](#), on page 972).

Figure 263: Station screen

change station 75001	STATION	Page 2 of X
FEATURE OPTIONS		
LWC Reception? 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	Restrict Last Appearance? y	
Bridged Call Alerting? n		
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:	Select Last Used Appearance? n	
AUDIX Name:	Coverage After Forwarding? _	
Recall Rotary Digit? n	Multimedia Early Answer? n	
Remote Softphone Emergency Calls: as-on-local	Direct IP-IP Audio Connections? n	
Emergency Location Ext: 75001	IP Audio Hairpinning? n	

- 2 In the **Per Button Ring Control** field, perform one of the following actions:
 - Type **y** if you:
 - Want users to select ringing individually for each call appearance, bridged call appearance, or analog bridged call appearance on the telephone, and
 - Want to enable the automatic abbreviated and delayed ring transition for each call appearance on the telephone, and
 - Do not want the system to automatically move the line selection to a silently alerting call, unless that call was audibly ringing earlier
 - Type **n** if you want:
 - Calls on call-appr buttons to always ring the telephone, and
 - The value in the **Bridged Call Alerting** field of the **Station** screen to control whether calls ring on the brdg-appr or the abrdg-appr buttons, and
 - The system to move the line selection to a silently alerting call, if there is no call audibly ringing the telephone
- 3 Press **Enter** to save your changes.

Assigning an abbreviated ringing button to a user

To assign an abbreviated and delayed feature button to a user:

- 1 Type **change station *n***, where *n* is the number of the extension to which you want to assign an abbreviated and delayed feature button for a user. Press **Enter**.

The system displays the [Figure 264, Station screen](#), on page 973).

Figure 264: Station screen

change station 32009		Page 3 of 4
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	7:	
2: call-appr	8:	
3: call-appr	9:	
4: abrv-ring Ext:	10:	
5:	11:	
6: headset	12: release	

- 2 Page through the screens until you see the **BUTTON ASSIGNMENTS** area.
- 3 Type **abrv-ring** next to the button that you want the user to use to cause a call that rings to transition from ringing to silence, or silence to ringing.
When you type **abrv-ring** next to the button, the system displays an **Ext** field.
- 4 (Optional) In the **Ext** field, type the extension of the other user to whom you want to assign the abbreviated ringing button.
- 5 Press **Enter** to save your changes.

Reports for Ringling - Abbreviated and Delayed

The following reports provide information about the Ringling - Abbreviated and Delayed feature:

- None

Considerations for Ringling - Abbreviated and Delayed

This section provides information about how the Ringling - Abbreviated and Delayed feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Ringling - Abbreviated and Delayed under all conditions. The following considerations apply to Ringling - Abbreviated and Delayed:

- You cannot assign Ringling - Abbreviated and Delayed to an attendant console.
- You can assign the Ringling - Abbreviated and Delayed feature to analog telephones. However, because analog telephones cannot visually alert, a user can unexpectedly answer an incoming call, when the user intends to originate a call.

Interactions for Ringling - Abbreviated and Delayed

This section provides information about how the Ringling - Abbreviated and Delayed feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Ringling - Abbreviated and Delayed in any feature configuration.

- **Call Coverage**

If the number-of-rings interval for coverage is shorter than the automatic transition interval, the system redirects the call to coverage before the system audibly alerts a call appearance that has delayed ringing. However, the system continues to increment the timer for the automatic transition interval, in case no coverage point is available, and the call continues to alert at the telephone.

When a call is immediately redirected to coverage, Abbreviated and Delayed ringing has no effect on the system processes.

- **Call Forwarding - Busy/Don't Answer**

When the system forwards a call because no one answers the call in the specified interval, the call stops alerting at the telephone. The Ringling — Abbreviated and Delayed feature does not affect the manner in which the system processes the call. However, the system continues to increment the timer for the automatic transition interval, in case forwarding fails, and the call continues to alert at the telephone.

If the interval for call forward don't answer is shorter than the interval for automatic transition, the system redirects the call to the forwarded-to extension before the call rings at a telephone that has a ring type of delayed ringing.

- **Call Vectoring - Expert Agent Selection - Logical Agents**

Calls that the system routes to a logical agent use the translations for the Ringling - Abbreviated and Delayed feature of the telephone that the agent uses.

- **Data Extension Calls**

Data Extension calls are not affected by the ring values. The calls continue to be directed according to the way that you administered bridged call alerting.

- **Hospitality Features - Do Not Disturb**

The Do Not Disturb feature takes precedence over the Ringling - Abbreviated and Delayed feature in blocking ringing to the telephone.

- Integrated Services Digital Network (ISDN) - World Class Basic Rate Interface (BRI)
Several of the protocol variations that the World Class BRI feature supports do not allow the messaging required for control of a telephone ringer by the Ringling - Abbreviated and Delayed feature. If the protocol variations do not permit the required messaging for the telephone ringer, the system rings the call at the telephone, and does not transition the ring.
- Multiappearance Preselection and Preference
If the **Per Button Ring Control** field on the **Station** screen is set to **n**, the system automatically selects any call that alerts at a telephone. The call can alert in a manner other than ringing.
If the **Per Button Ring Control** field on the **Station** screen is set to **y**, the system automatically selects any call that rings at a telephone.
- Off-Premises Station (OPS) and Off-Premises Extension (OPX) lines
You must use a ring type of **ring** for OPS and OPX lines.
- Personal Central Office Line (PCOL) calls
Ring values do not affect the processing of PCOL calls. PCOL calls continue to be directed according to the way that you administer bridged call alerting.
- Redirection Notification
If you enable Redirection Notification, telephones receive redirection notification only if the alerting button, or the first call appearance, has an assigned ring value of **ring** or **abrv-ring**.
- Terminating Extension Group (TEG) Calls
Ring values do not affect the processing of TEG calls. TEG calls continue to be directed according to the way that you administer bridged call alerting.
- Voice mail systems
Voice mail systems might look for ringing that is applied to a port to trigger call answer. Ring-type translations that are inappropriately set for ports that serve a voice mail system, can result in undesirable operation of the adjunct.

Security Violation Notification

Use the Security Violation Notification (SVN) feature to notify a designated referral point about a possible security violation. A designated referral point 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 Communication Manager 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.

Detailed description of Security Violation Notification

This section provides a detailed description of the Security Violation Notification (SVN) feature:

- Security violation thresholds and notification
- Sequence of events
- Reporting
- SVN - halt buttons
- SVN Referral Call With Announcement
- Dealing with Security Violations

To effectively monitor the security of your system, you must 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:

Avaya recommends that you print and clear the security-violation measurement reports at least once a month. In a busy system, you must print security-violation measurement reports often.

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 server running Communication Manager.
- 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**. Press **Enter**.

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 media server or switch), SVN referral calls with announcements can terminate to a point on- or off-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.

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 Avaya Communication Manager using a login ID with the correct permissions.
- 2 Type **disable login *n***, where *n* is the login ID of the user. Press **Enter**.

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 Avaya Communication Manager using a login ID with the correct permissions.
- 2 Type **enable login *n***, where *n* is the login ID of the user. Press **Enter**.

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 Avaya Communication Manager using a login ID with the correct permissions.
- 2 Type **enable remote-access**. Press **Enter**.

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 Avaya Communication Manager using a login ID with the correct permissions.
- 2 Type **disable remote-access**. Press **Enter**.

Hardware requirements for Security Violation Notification

The Security Violation Notification (SVN) feature requires the following hardware:

- None

Administering Security Violation Notification

The following steps are part of the administration process for the Security Violation Notification (SVN) feature:

- [Setting up Security Violation Notification](#)

This section describes:

- Any prerequisites for administering the Security Violation Notification (SVN) feature
- The screens that you use to administer the Security Violation Notification (SVN) feature
- Complete administration procedures for the Security Violation Notification (SVN) feature

Prerequisites for administering Security Violation Notification

You must complete the following actions before you can administer the Security Violation Notification (SVN) feature:

- Ensure that the `Access Security Gateway` field on the **Optional Features** screen is set to `y`. If the `Access Security Gateway` field is set to `n`, your system is not enabled for this feature. Contact your Avaya representative before you continue with this procedure.

To view the **Optional Features** screen, type `display system-parameters customer-options`. Press **Enter**.

Screens for administering Security Violation Notification

Screen name	Purpose	Fields
Optional Features	Ensure that the system is enabled for this feature.	<code>Access Security Gateway</code>

Screen name	Purpose	Fields
<i>Security-Related-System Parameters</i>	Set up security violation notification.	• SVN Login Violation Notification • Originating Extension • Referral Destination • Login Threshold • Time Interval
<i>Remote Access</i>	Set up security violation notification.	• Disable Following A Security Violation
<i>Station</i>	Set up security button assignments.	

Setting up Security Violation Notification

To set up security violations notification:

- 1 Type **change system-parameters security**. Press **Enter**.

The system displays the *Security-Related System Parameters* screen ([Figure 265, Security Related System Parameters screen](#), on page 981).

Figure 265: Security Related System Parameters screen

change system-parameters security	Page 1 of 2
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:

If you are not using Remote Access, go to Step 11.

- 8 (Optional) Type **change remote-access**. Press **Enter**.
The system displays the **Remote Access** screen ([Figure 266, Remote Access screen](#), on page 982).

Figure 266: Remote Access screen

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 (Optional) In the **Disable Following A Security Violation** field, type **y**.
This disables Remote Access following detection of a remote access security violation.
- 10 (Optional) Press **Enter** to save your changes.
- 11 Type **change station n**, where **n** is the station to be assigned the notification halt button. Press **Enter**.
The system displays the **Station** screen ([Figure 267, Station screen](#), on page 983).

Figure 267: Station screen

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.

Reports for Security Violation Notification

The following reports provide information about the Security Violation Notification feature:

- The Security Violations Status Report shows details of the last sixteen violations of each type. The Barrier Code and Authorization Code reports also include the calling party number from which the attempt was made, if that information is available.

Considerations for Security Violation Notification

This section provides information about how the Security Violation Notification feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Security Violation Notification under all conditions. The following considerations apply to Security Violation Notification:

- 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 for Security Violation Notification

This section provides information about how the Security Violation Notification feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Security Violation Notification in any feature configuration.

- 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.

Separation of Bearer and Signaling

Use the Separation of Bearer and Signaling feature to reduce the costs of private, leased lines.

Detailed description of Separation of Bearer and Signaling

This section provides a detailed description of the Separation of Bearer and Signaling (SBS) feature.

SBS provides a low-cost, “virtual” private network over IP trunks with the high voice quality that is expected of the public switched telephone network (PSTN). Thus, with SBS, you save on the costs of private, leased lines.

With SBS distributed communication system plus (DCS+) customers can replace an expensive virtual private network (VPN) service if the customer needs Dial Plan Expansion (DPE) functionality. DCS does not work with 6- or 7-digit dial plans. Although QSIG work with 6-digit or 7-digit dial plans, QSIG does not work over VPNs. Customers can get enhanced signaling without private leased lines.

SBS also provides a transport mechanism for application data, and other enhanced functionality in locations where ISDN trunking is unavailable or expensive.

The SBS feature supports:

- QSIG private network signaling over a low-cost, IP-based network
- Voice calls (bearer traffic) over the PSTN
- Correct association between QSIG feature signaling information and each voice call.

You must always use Automatic Alternate Routing/Automatic Route Selection/Uniform Dial Plan (AAR/ARS/UDP) to originate an SBS call. You cannot use a Trunk Access Code or a Dial Access Code to originate an SBS call.

Hardware requirements for Separation of Bearer and Signaling

The Separation of Bearer and Signaling feature requires the following hardware:

- None

Administering Separation of Bearer and Signaling

This section describes the screens that you use to administer the Separation of Bearer and Signaling (SBS) feature.

Screens for administering Separation of Bearer and Signaling

Screen name	Purpose	Fields
<i>Feature-Related System Parameters</i>	Specify a valid country code for the SBS signaling trunk groups.	Local Country Code
	Specify the access code that the private switched telephone network (PSTN) requires to route calls out of the country.	International Access Code
<i>ISDN Numbering - Public/Unknown</i>	Specify information for ISDN processing.	All
<i>Signaling Group</i>	Enable SBS for a signaling group.	SBS
<i>Station</i>	Specify the SBS Extension.	SBS Extension
<i>Trunk Group</i>	Enable SBS for a trunk group.	SBS

Reports for Separation of Bearer and Signaling

The following reports provide information about the Separation of Bearer and Signaling feature:

- None

Considerations for Separation of Bearer and Signaling

This section provides information about how the Separation of Bearer and Signaling (SBS) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Separation of Bearer and Signaling under all conditions. The following considerations apply to Separation of Bearer and Signaling:

- Call Detail Recording (CDR)

CDR functionality records the calling and called number information, and the start and end time of each measured call. You administer this CDR functionality on a trunk group basis within Avaya Communication Manager. At the end of a call, the information is made available to an adjunct server for various functions, such as costing, reports, and traffic analysis.

In the case of an SBS call, the data that is gathered and made available to the adjunct server does not change. However, since each trunk group that is involved in an SBS call can be administered to generate CDR reports, the system can generate two CDR records for each SBS call. The system generates one record for the SBS signaling call, and one record for the SBS bearer call. *No way exists to link the separate SBS signaling and bearer CDR records.*

Since the signaling and bearer CDR records cannot be linked, it might be preferable to measure only the bearer calls. The service provider is more likely to bill for the bearer calls. However, note that the SBS bearer call is always answered by the SBS extension, even though the actual called party does not answer the SBS call. Also note that the parties on the SBS bearer call are not the actual originating and terminating parties. The parties on the SBS bearer calls are “dummy” users, who are internal to Communication Manager for the sole purpose of originating and terminating the bearer call.

- Call Management System (CMS) and Basic Call Management System (BCMS)

You can administer the following entities to be measured for CMS or BCMS:

- A trunk group
- A Vector Directory Number (VDN)
- A hunt group

An SBS call can generate two separate CMS or BCMS records. The system can generate one record for the SBS signaling call, and one record for the SBS bearer call.

CMS and BCMS measure the SBS bearer call only if the bearer trunk group is administered to be measured. CMS and BCMS measure only the trunk bearer seize events and idle events.

The SBS signaling call is measured if either the SBS signaling group is administered to be measured, or if an endpoint on the SBS call is a measured object, such as an agent in a hunt group that is administered to be measured. The events that are measured on the SBS signaling call include not only the trunk signaling seize events and idled events, but also any endpoint events, such as agent hold.

The SBS signaling and bearer calls generate separate CMS or BCMS records with different Universal Call IDs (UCIDs). *No way exists to link these separate records.*

Interactions for Separation of Bearer and Signaling

This section provides information about how the Separation of Bearer and Signaling (SBS) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Separation of Bearer and Signaling in any feature configuration.

Overview

SBS provides a new type of trunk call, a separated trunk call. Many types of trunks can be used to carry the bearer portion of the separated call. Thus, SBS affects many Communication Manager features. Almost any feature that can interact with a trunk call, can interact with SBS.

This section describes features that might interact with SBS calls. Other features might also interact with SBS in a minor way.

You must understand the way SBS calls are configured internally to be able to understand why certain features can work with SBS calls, and others cannot. This general understanding also helps to accurately predict whether a feature that is not included in this section can successfully work with an SBS call.

Two important facts about the internal configuration of an SBS call are:

- The SBS bearer and SBS signaling calls are tracked internally as separate calls for the life of the SBS call. Separate call records are created internally for each call to:
 - Allow the SBS bearer call to be carried over almost any type of trunk
 - Use the standard call establishment and tear down code for that trunk type

The system cannot use the standard code if the bearer trunk is somehow “buried” as a nonstandard party in a “merged” signaling and bearer call record.
- The endpoint users are parties on the SBS Signaling call, not on the SBS bearer call. In other words, the SBS signaling call controls call. Since endpoint user activity drives QSIG signaling only if a QSIG trunk is a party on the call, the SBS Signaling call must control the call. A QSIG trunk is guaranteed only if the endpoint user is associated with the SBS trunk, not if the endpoint user is associated with the bearer trunk.

Understanding potential interactions

With these two facts about the internal configuration of an SBS call, the following general observations about feature interactions with SBS calls can be made.

- Most features work with SBS

A significant number of the features that can interact with an SBS call, work as the features do with any other QSIG trunk call.
- Delay applies to all SBS calls

For features that do work with SBS calls, the standard SBS delay applies. However, some the following feature interaction descriptions do not specifically mentioned SBS delay.
- Call status

Any query or report that is related to an endpoint user on an SBS call indicates the call ID of the SBS signaling call, and the trunk ID of the signaling, not the bearer, trunk. For example, a **status station** command, that is issued against a station on an SBS call, shows the station that is connected to an SBS trunk.
- Feature signaling interworks to/from SBS Signaling calls

Any feature information that is interworked from a non-SBS leg of a call to an SBS leg of a call is transported on the SBS signaling call, not on the SBS bearer call. Likewise, any feature information interworked from an SBS leg of a call to a non-SBS leg of a call is taken from the SBS signaling call, not from the SBS bearer call.
- Two records per SBS call

Any feature that reports or records status for a call can create two different reports or records for a single SBS call. The feature can create one report or record for the signaling trunk, and a second report or record for the bearer trunk. For example, with Call Detail Recording (CDR), the result depends on whether both trunks groups are administered to produce call detail records.
- Bearer trunk signaling features

Features that require signaling over a non-QSIG type trunk do not work with SBS calls. The endpoint users are not parties on the SBS bearer call, and so user activity cannot drive any feature signaling on the bearer call. Similarly, any feature signaling that is received on the bearer call cannot drive any notification or displays to the endpoint user. For example, public or private network-specific (non-QSIG) Malicious Call Trace (MCT) network notification and Advice of Charge (AOC) display functionality do not work with SBS calls.

- Bearer trunk user features

Features that are related to the SBS bearer call that require activation or acknowledgement from an endpoint user do not work with SBS calls. Such features do not work with SBS calls, because the endpoint user is not a party on the bearer call. For example, queuing of the SBS bearer call does not work because the real originating party is not on the bearer call.

- Early answer features

Features that require early answer, for example, to pass tones, do not work with SBS calls. Such features do not work with SBS calls, because when the signaling call is answered, the bearer call is not started. When the bearer call is first answered, the call is for the SBS extension at the terminating node. For example, authorization code collection on incoming calls and direct calls to remote access does not work with SBS calls.

- Network features that send tones when the bearer call answers do not work

Network features that send tones when the bearer calls do not work with SBS. When the bearer call is answered, the call is for the SBS extension. The final endpoint user is not on the call to hear the tones. For example, the DTMF notification when a network call is eligible to be transferred, such as with Take-back and Transfer, is not heard.

General system features

- Media processor resources are not used by the SBS trunks, which carry only SBS signaling calls. SBS bearer calls require media processor resources if IP trunks are used.
You can encounter voice quality degradation if IP trunks are used for bearer calls.
- Shuffling and hairpinning work with SBS bearer calls if IP trunks are used. The SBS bearer call originates with the same “shufflable endpoint” characteristics as the real originator.
- The contents of the Incoming Call Identification (ICI) display for an SBS call is obtained from the SBS signaling call, not the bearer call.
- The Class of Restriction (COR) of the SBS call originator is used to set up the SBS bearer call.
- The system routes the SBS signaling and bearer calls separately. Use caution when you administer Toll Restriction, Toll Analysis, and Toll/Code restriction, so that you do not block calls that should be allowed.
- The Class of Service (COS) Trunk-to-Trunk Transfer permission affects the transfer of SBS trunk calls in the same way as for non-SBS trunk calls.
- SBS calls follow the Station Hunting of the originally called party.
- When Tenant Partitioning is active in an enterprise, existing Tenant Partitioning rules apply to endpoints, SBS extensions, SBS trunk groups and the bearer trunk groups that are involved in any SBS call.
- SBS works with Dial Plan Expansion of 6-digit or 7-digit extensions.
- Standard Malicious Call Trace (MCT) on an incoming SBS call records the bearer trunk call.

Attendant

Attendant features that do not work with SBS

- Attendant Control of Trunk Group Access does not work with an SBS Trunk Group.
- Centralized Attendant Service (CAS) does not work over SBS trunks, since CAS requires release link trunks (RLT). However, you can use CAS to direct an incoming SBS call to a centralized attendant over an RLT using CAS. Likewise, a centralized attendant can extend a call over an RLT, that the system then routes to an SBS trunk.

Attendant features that work with SBS

- Attendant Direct Extension Selection (DXS) can be used to originate an SBS call that uses Uniform Dial Plan (UDP).
- Attendant Intrusion can be used to intrude on an SBS call.
- Attendant Recall works with an SBS call
- Attendant Return Call and Serial Calling work with an incoming SBS call.
- Attendant transfer is applicable to an SBS call.
- An incoming SBS call can be parked, and is subject to Call Park Time-out to the Attendant.
- An SBS call can be a party in an attendant conference.
- An SBS call can be directed to an individual attendant access number.
- Inter-PBX attendant service works with SBS trunks.
- Incoming SBS calls follow attendant Night Service.
- Incoming SBS calls hear Recorded Announcements in the attendant queue.
- An incoming SBS call can be answered using Trunk Answer Any Station (TAAS).
- The Trunk Identification by Attendant feature identifies the SBS trunk group member.
- An SBS call can be held with Two-Party Hold on Console.
- Attendant Vectoring can receive and redirect attendant-directed calls over SBS trunks.

Adjunct Switch Applications Interface (ASAI)

- When you use Call Classification after Answer, do not use SBS to route calls. Do not use SBS because the Call Classifier must be on the bearer trunk, and the SBS-invoked DTMF signaling that is sent on the bearer trunk causes interferences.
- ASAI Phantom call, such as DEFINITY Anywhere, can originate or receive an SBS call.
- ASAI Selective Listening works on an SBS trunk party on a call.
- ASAI Send DTMF works on a connected SBS call.
- ASAI Single Step Conference can add another station onto a call with an SBS call.
- ASAI Provided Dial-Ahead Digits work when the incoming call is an SBS call.
- Any ASAI user data, Universal Call ID, or both that are currently transported or interworked over QSIG trunks, are sent on the SBS signaling call.

- When an SBS call is received and subsequently tandemed over an SBS trunk, any II Digits that are received in SETUP message to the incoming SBS Signaling call, are tandemed with the SBS signaling call.
- The information that is provided in response to an ASAI Value Query indicates whether an SBS trunk or an associated bearer trunk is idle or busy. The response is reported based on the uid in the query message, and whether it was for the signaling or the bearer trunk.
- For ASAI Event Reports, Communication Manager reports the Call ID and the Trunk ID of the SBS trunk. ASAI should provide the Trunk ID of the associated bearer trunk to the ASAI Event Reports and Adjunct Route message.

Intuity AUDIX and Octel voice mail adjuncts

- Centralized voice mail with Interswitch Mode Codes does not interwork with SBS trunks. For that application, the tie trunks between the servers that run Communication Managers cannot use the QSIG protocol. While those trunks might be used for SBS bearer calls, it is unlikely that when this methodology is implemented that SBS is also implemented.
- Leave Word Calling with Message Wait Indicator over QSIG (QSIG LWC MSI) supports Digital Line Emulation integration for centralized voice mail using SBS.
- SBS supports Intuity centralized Intuity AUDIX from a served user switch works over SBS.
- Where an Octel Serenade is connected to Communication Manager with QSIG, the server that runs Communication Manager is the SBS terminating node that interworks to the Serenade, since the Serenade does not support SBS.

Call Center

Automatic Call Distribution (ACD)

- Look Ahead Interflow (LAI) might not function correctly on an SBS call, because of the SBS call setup delay. If the SBS delays are a problem, you might try polling by Best Service Routing (BSR) using Non Call Associated (NCA) Temporary Signaling Connection (TSC) instead of LAI.
- If Outbound Calling is done from a Call Center, and the call uses SBS to call another Communication Manager system, the call is affected by the normal SBS call setup delays. Outbound Calling over SBS trunks must not use Call Classification, since there will be interference from the DTMF signaling invoked by SBS and sent to identify the correct bearer call at the terminating end.
- Dialed Number Information Service (DNIS) and Original Dial Number Delivery service from a service provider can deliver an SBS bearer call to an SBS extension in an SBS terminating node.
- When an incoming ACD call arrives via an SBS trunk, transfers by an agent to another agent or to an application work properly.
- An incoming SBS call can hear any announcements that are associated with the Call Center.
- The agent Assist functionality works when the incoming call arrives over SBS.
- Displays at the agent terminals function correctly when the incoming call arrives over SBS. This includes Vector Collected digits.
- Multiple Call Handling works when arriving calls are incoming SBS calls.
- An incoming SBS call that receives Redirect on No Answer (RONA) works correctly.

- SBS calls that receive intra-flow, inter-flow or hunt group night service treatment are routed in the same way as a non-SBS call.
- Any ISDN and or R2MFC call data, that is currently interworked to or from a QSIG trunk is sent on (retrieved from) the SBS signaling call at the SBS interworking node. Such call data includes calling party number, II digits, CINFO digits, and so on.
- Call Center data that is currently transported on QSIG trunks is sent on the SBS signaling call.
- When an ACD call is transferred to an agent via an SBS trunk connection on another system that runs Communication Manager, all associated call information that is currently transported on a QSIG trunk is sent on the SBS signaling call.
- All types of Service Observing functions work on SBS calls in the same way as for non-SBS calls.

Best Service Routing

- Polling by Best Service Routing (BSR) can use SBS signaling facilities if the NCA-TSC version of BSR Polling is used, and adequate resources are available. The non-NCA-TSC version of BSR Polling might not work because of the SBS call setup delays.
- BSR interflow over SBS trunks, including incoming call data forwarding and Enhanced Information Forwarding, are supported.

Vectors

- Routing on Automatic Number Identification (ANI) by a vector can use SBS to route an outgoing call. For an incoming SBS call, the Routing on ANI functionality uses the ANI from the SBS signaling call.
- Correct routing over SBS trunks works when a vector step exists for routing on ANI or II digits, or a route-to number step and interflow is invoked.
- Post-Connect in-band DTMF signaling for Call Prompting collect steps and Auto Attendant functionality both work.
- When CINFO Digit Routing occurs, and the call is routed over an SBS trunk, the information is tandemmed with the call.
- Vector Routing Tables can use SBS trunks to route calls.
- Incoming SBS calls still provide VDN of Origin announcements (VOA) and displays to the agent who answers the call.
- VDN Return Destination works with SBS calls.

Networking Related Interactions

Networking features or capabilities that do not work

- Authorization Codes cannot be collected on an incoming SBS signaling or an SBS bearer call (for example, when required through administration of the incoming trunk group or when required because of insufficient facilities restriction level (FRL) on a tandem call). Authorization codes cannot be collected on incoming SBS signaling calls because the SBS bearer call is not established at the time the system usually prompts for and signals an authorization code. Also, authorization codes cannot be collected on incoming SBS bearer calls because the originating endpoint user is not a party on the SBS bearer call.

- When the bearer call is transported through Message Oriented Signaling trunks ISDN, information in the SBS bearer call D-channel is not displayed to the end users. The information is not displayed because the signaling in the SBS signaling call overwrites the information.
- User information that is in the SBS bearer leg of the call is ignored.
- Australian Malicious Call Trace (MCT) cannot be invoked on an SBS call. This is because end-user activity drives feature signaling only on the SBS signaling call, not on the SBS bearer call, and an SBS trunk does not support the Australia public network protocol.

The normal MCT feature within Communication Manager records the call as usual.

- ETSI MCT can not be invoked on an SBS call. End-user activity drives feature signaling only on the SBS signaling call, not on the SBS bearer call, and an SBS trunk does not support ETSI protocol.

The normal MCT feature within Communication Manager records the call as usual.

- Calling Line ID Prefix information is ignored when the information is transported in conjunction with an associated bearer trunk.
- Advice of Charge (AOC) information that is received on an SBS bearer call is not displayed to the end user. However, AOC information is conveyed to the CDR port for the SBS bearer call record. Only SBS trunk information affects end-user displays.
- ETSI Network Call Deflection (NCD) does not work with an SBS call. For SBS, the signaling that controls the call is in the SBS signaling call on the QSIG interface. NCD, however, requires an ETSI interface that is available only on SBS bearer calls.
- ETSI Network Call Transfer (NCT) does not work with an SBS call. For SBS, the signaling that controls the call is in the SBS Signaling call on a QSIG interface. NCT, however, requires an ETSI interface that is available only on the SBS bearer call.
- Direct SBS calling into Remote Access does not work. Barrier Codes cannot be collected for SBS calls to Remote Access because the SBS bearer call has not yet been established when such tones are prompted for or expected. However, an SBS call can invoke Remote Access by means of a vector collect or a route-to command.
- SBS does not support Wideband Switching (NxDSO).
- Russian Incoming ANI with a button does *not* display the ANI that is received in the SBS bearer call. Only the ANI received on the SBS signaling call is displayed to the end user.
- Trunk Flash to get recall dial tone from a central office (CO) does not work. End-user activity drives signaling on the SBS signaling call only, not the SBS bearer call. QSIG Call Transfer functionality can be used instead.
- R2 MultiFrequency Compelled (MFC) Intercept treatment must drop the SBS bearer and SBS signaling calls, and does so by applying the appropriate treatment to the call originator based on what is received on the R2 MFC bearer call.

Networking features or capabilities that work

- Authorization codes can be collected when required on an outgoing SBS call, that is, when required to access the SBS trunk, since they are collected locally before the outgoing trunk is seized.
- Calling Party Number (CPN) restriction can be administered for and signaled on both the SBS signaling and SBS bearer calls. However, end-user displays, including any restrictions, are populated from information that is carried in the SBS signaling call, not from information in the SBS bearer call.

- At an SBS Interworking node, DCS and DCS+ signaling on the non-SBS portion of the call is interworked to and from the SBS signaling call, to the extent that DCS-QSIG interworking currently applies.
- A DCS trunk can be used as the bearer trunk on an SBS call. However, any DCS signaling information that is received on the Bearer call is overridden by the QSIG signaling information received on the SBS signaling call.
- User information that is received on an SBS signaling call by an SBS tandem node, or received on a non-SBS trunk by an SBS interworking node, is sent on the SBS signaling call, per current tandeming or interworking procedures on QSIG trunks.
- Temporary Signaling Connection (TSC) messages can be sent over an SBS trunk.
- Look Ahead Routing can be used with both SBS signaling and SBS bearer calls, if such calls start on ISDN trunks.
- In regions where Feature Plus (F+) is offered, SBS calls can use the pseudo-DID functionality of F+ to avoid the need to obtain DID or DDI numbers from a service provider. You must administer SBS extensions at the SBS terminating node, but these numbers do not need to correspond with real DID or DDI numbers. CPN prefix administration at the SBS terminating node must map the SBS extension to a number that is a national complete number, except that the SBS extension portion is not recognized by the PSTN. ARS at the SBS originating node must route this number to a route pattern preference that supports F+. The “No. Dgts SubAddress” administration for this preference must indicate the length of the SBS extension at the SBS terminating node. This number is the number of digits that are extracted from right to left, and sent in the Calling Party Subaddress Information Element. Administration for this preference must also delete the SBS extension digits, and insert the Listed Directory Number (LDN) extension of the SBS terminating node in its place. The SBS bearer call is routed to the LDN at the SBS terminating node. F+ functionality at the SBS terminating node then routes the call to the SBS extension, passed in the Subaddress IE, instead of to the attendant. Multiple route patterns are needed at the SBS originating node, if the SBS terminating node uses SBS extensions of various lengths.

SBS extensions used with F+ cannot be longer than five digits, because F+ is not included in Dial Plan Expansion.
- QSIG MSI messages are sent in the SBS Signaling link. The system ignores any messages that are in the SBS bearer call.
- QSIG Call Completion works with SBS calls.
Both the original call, and the “call-back call” incur separate SBS delays if the call uses SBS trunks.
- QSIG Call Transfer works with SBS calls.
Both the original call and the second call, to the transferred-to party, incur separate SBS delays if the call uses SBS trunks.
- QSIG Diversion, forward switch and reroute, works with SBS calls.
Both the original call and the second call, to the forwarded-to party, incur separate SBS delays if the call uses SBS trunks.
- QSIG Path Replacement works with SBS calls.
The entire SBS call, both the SBS signaling call and the associated SBS bearer call, are replaced. All separate calls, the original call, the call to the transferred-to party, and then the path replacement call, incur separate SBS delays if the calls use SBS trunks.
- QSIG Enhanced Path Replacement works with SBS calls. Multiple SBS delays apply.

- Non-Avaya QSIG MSI are tandemed to and from any non-SBS QSIG portion of an SBS call (at the SBS interworking node) and on the SBS signaling call (at an SBS tandem node) per existing QSIG transit operation.
- QSIG MWI works with SBS calls.
- QSIG Temporary Signaling Connections (TSCs), known as Call Independent Signaling Connections (CISCs) in QSIG literature, are supported. Non Call Associated Temporary Signaling Connections (NCA TSCs) are signaling-only connections that transport feature information. While NCA TSCs can be initiated as a result of some activity on a bearer call, NCA TSCs are independent of bearer calls. NCA TSCs are set up as nonbearer calls, and use a call reference value (CRV) that is different than any CRV that is in use on any other existing bearer or signaling call on that interface.

Communication Manager supports two different NCA TSC protocols. Use the signaling group **Supplementary Service Protocol** field on the **Signaling Group** screen to administer these protocols. The **Supplementary Service Protocol** field is set to **a** for AT&T NCA TSCs, and to **b** for QSIG NCA TSCs (CISCs).

For full QSIG functionality, you must set the **Supplementary Service Protocol** field on both the **Trunk Group** screen and the **Signaling Group** screen to **b** (QSIG). You must set this field to **b**, because some QSIG features, such as QSIG Call Completion and QSIG Message Waiting Indication, use QSIG feature signaling on both the bearer call and on an NCA TSC to work properly.

- QSIG Centralized Attendant Service with MSI works with SBS calls.
- QSIG transit capabilities are supported with SBS calls through tandeming of QSIG signaling to and from any non-SBS QSIG portion of an SBS call (at the SBS interworking node), and on the SBS signaling call (at an SBS tandem node), per existing QSIG transit operation.
- QSIG VALU signaling works with SBS calls. You might need to increase the timer that is used to return QSIG VALU Call Coverage calls back upon no answer, so that SBS delays do not cause such calls to be returned prematurely. Use the **Local Cvg Subsequent Redirection/CFWD No An Interval (rings)** field on the **System Parameters Call Coverage/Call Forwarding** screen to administer the timer.
- QSIG Called/Busy Name ID is supported in the SBS signaling call.
- QSIG Calling/Connected Name/Number ID is supported in the SBS signaling call.
- QSIG Call Offer is supported in the same way as over normal QSIG trunks.
- Automatic Alternate Routing (AAR), Automatic Route Selection (ARS), or both AAR and ARS can be used to route the SBS signaling call. SBS bearer calls are routed by ARS only, but also can be directed to AAR from ARS.
- SBS calls to an analog station endpoint display the SBS signaling information if the endpoint is served by a TN793 or TN2793 port.
- The Russian Transit/Power Industry Tie Trunk is expected to work as an associated bearer trunk.
- The X-Station Mobility feature works for incoming or outgoing calls that are routed with SBS.
- Leave Word Calling (LWC) for Unanswered External Calls with Automatic Number Identification (ANI) is supported with the information in the SBS signaling call that is stored for the called party.
- ISDN Feature Plus calls are supported, as the SBS bearer call, with the SBS signaling call information that is used for endpoint displays.
- ISDN Calling Party Number Presentation options are supported by SBS in the SBS signaling call.

Separation of Bearer and Signaling

Interactions for Separation of Bearer and Signaling

- QSIG/DCS Partial Reroute works is the same way as it does currently, with the SBS signaling call as the QSIG part of the call.
- DS1 With Echo Cancellation is supported for the SBS bearer call.

Service Observing

Use the Service Observing feature to allow designated users to listen to another user call.

Detailed description of Service Observing

This section provides a detailed description of the Service Observing feature.

This section describes service observing in environments without Automatic Call Distribution (ACD) or call vectoring. See *Avaya Communication Manager Contact Center Guide to ACD Contact Centers* to use service observing in ACD or call vectoring environments.

With Service Observing, designated users, usually supervisors, can listen to other user calls. The user that observes the calls of another user is called an *observer*. Use the Service Observing feature to train agents, or to monitor the quality of service in call centers and other environments where employees serve customers over the telephone.

WARNING:

Listening to the call of another user can be subject to federal, state, or local laws, rules, or regulations. You might need to obtain the consent of one or both of the parties on the call. Ensure that you know, and comply with, all applicable laws, rules, and regulations when you use this feature.

An observer can monitor calls to any of the following entities:

- An extension
- A vector directory number (VDN), on systems with call vectoring
- A logical agent ID, on systems with Expert Agent Selection (EAS)

Observers can monitor calls in listen-only mode or listen-and-talk mode. In listen-and-talk mode, an observer can hear and speak with all parties on a call. The user that is monitored does not know that an observer listens to the call, unless you administer Avaya Communication Manager to provide a monitoring tone.

When an observer is off site, the observers can use remote access to monitor calls. In systems with call vectoring, a vector can control access to Service Observing.

If an observer uses a telephone that has a service observe button, the button:

- Blinks while the observer waits for an eligible call
- Lights steadily while the observer observes a call

The system does not reserve a call appearance while the observer is in the wait state, if:

- The observer uses a feature access code (FAC) to activate Service Observing
- No service observe button is administered for the telephone

An idle call appearance must be available for an observer to go to the observing state when an eligible call arrives.

Listen and talk modes

When an observer uses the feature button for Service Observing, the observer can toggle between listen-and-talk mode and listen-only mode. However, when an observer activates Service Observing with an FAC, the observer must choose the listen-only mode or the listen-and-talk mode at the start of the session. If the user wants to change modes after the session starts, the observer must end the session, and then chose the other mode when the observer starts a new session. The FACs for Service Observing are the

- Service Observing Listen Only Access Code
- Service Observing Listen/Talk Access Code

Note that the system also requires an FAC for remote Service 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.

Restrictions

Two observers cannot monitor the same extension, or the same call, simultaneously. The system generates the busy signal for the second observer that attempts to use Service Observing for a call.

If two users are being observed independently, and one of the users calls the other user, the observer of the calling extension observes the call. The observer of the called extension goes into wait state until the call is over.

Telephone displays

The system displays the same information for both the user and the local observer. The system adds the text “so,” for Service Observing, to the display of the observer.

Trunk calls

When a user places a trunk call, Service Observing starts when the user finishes dialing the call. For calls on central office (CO) trunks, the system considers dialing to be complete when the answer supervision is returned, or when the answer supervision timeout occurs.

The system denies any attempt to use Service Observing over trunks that do not have disconnect supervision.

Warning and conference tones

You can administer a tone that notifies the parties on a call that the call is observed. You can administer the tone as a warning tone or a conference tone.

If you administer the:

- Warning tone the system generates a unique 2-second, 440-Hz tone before an observer connects to the call. While the call is observed, the system repeats a shorter version of this tone every 12 seconds. If you administer the
- Conference tone, the system generates the conference tone before an observer connects to the call. The system does not repeat the conference tone during the call.

Hardware requirements for Service Observing

The Service Observing feature requires the following hardware:

- None

Administering Service Observing

This section describes the screens that you use to administer the Service Observing feature

Screens for administering Service Observing

Screen name	Purpose	Fields
<i>Class of Restriction</i>	Set up the Class of Restriction (COR) to support Service Observing.	• Can Be Service Observed • Can Be Service Observer • Service Observing COR Table
<i>Feature Access Code (FAC)</i>	Specify the FACs for Service Observing.	• Service Observing Listen Only Access Code • Service Observing Listen/Talk Access Code

Screen name	Purpose	Fields
Feature-Related System Parameters	Enable Expert Agent Selection (EAS) for the system.	Expert Agent Selection (EAS) Enabled?
	Enable a conference tone for observed users.	Service Observing Conference Tone?
	Enable a warning tone for observed users.	Service Observing Warning Tone
Station (multiappearance)	Assign the feature button for Service Observing.	Button/Feature Button Assignment (basic, VDN, Logical Agent ID) serv-obsrv
Optional Features	Enable Service Observing for: - Basic or logical agent ID observing - Remote access - Vector Directory Numbers (VDNs) - Vector-initiated observing	- Service Observing (Basic) - Service Observing (Basic) and Service Observing (Remote/By FAC) - Service Observing (Basic) and the Service Observing (VDNs) - Vectoring (Prompting)

End-user procedures for Service Observing

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Service Observing:

- Activating Service Observing
- Deactivating Service Observing

To activate Service Observing:

- 1 Press the service observing button or enter a feature access code (FAC).
- 2 Enter the extension that you want to observe.

When you use the service observing button to activate Service Observing, you start in listen-only mode. Press the service observing button to toggle between listen-only mode and listen/talk mode.

To deactivate Service Observing:

- Hang up, select another call appearance, or press the disconnect or release button.

Reports for Service Observing

The following reports provide information about the Service Observing feature:

- None

Considerations for Service Observing

This section provides information about how the Service Observing feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Service Observing under all conditions. The following considerations apply to Service Observing:

- None

Interactions for Service Observing

This section provides information about how the Service Observing feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Service Observing in any feature configuration.

- Attendants
An attendant can be observed, but cannot be an observer.
- Bridged Appearances
You can observe calls on primary extensions only, not on bridged appearances. For example, if you are observing extension 3082 and this telephone also has a bridged appearance for extension 3282. You cannot observe calls to extension 3282.
- Busy-Verification
You cannot observe an extension that is being busy-verified. You cannot busy-verify an extension that is being observed.
- Call Coverage/Call Pickup
An observer cannot observe a call that is 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 observed call.
- **Call Waiting**
Incoming calls do not wait on a single-line telephone that is being observed.
- **Conference**
An Observer cannot initiate a conference call while the observer is also observing a call.
If an observed user starts a conference, or enters a conference call that has fewer than six parties, the system places the observer in the wait state until the system connects the call. Then the observer can observe the conference. The system counts the observer as one of the conference call participants. The observer can observe all of the conference participants, regardless of the Class of Restriction (COR) In addition, the system bridges the observer onto any calls that a conference participant places or receives while the conference is active. When the user leaves the conference, the observer also leaves and returns to observing the original call.
- **Data Privacy**
An observer cannot observe an extension:
 - On which Data Privacy is active
 - While the extension is on a conference call with another extension for which Data Privacy is active
- **Data Restriction**
An observer cannot observe an extension
 - On which Data Restriction is active.
 - While the extension is on a conference call with another extension for which Data Restriction is active
- **Integrated Directory**
Observers do not hear a user dial an Integrated Directory number.
- **Distributed Communications System (DCS)**
To observe user extensions that are on another node, such as a DCS station extension, the observer must set up remote-access service observing.
The system does not transmit service observing displays across DCS networks.
- **Hold**
An observer cannot place a call on hold while the observer is also observing a call.
If a user who is being observed places a call on hold, the observer enters the wait state.
- **IP Solutions**
If an observer observes an IP-IP direct call, the users on the call might hear a break-in conversation of about 200 milliseconds.
- **Leave Word Calling (LWC)**
Parties on an observed call cannot use LWC.
- **Music-on-Hold**
If an observer is in listen-talk mode, neither the caller nor the 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

An observer cannot observe an extension:

- On which Privacy - Manual Exclusion is active.
- While the extension is on a conference call with another extension for which Privacy - Manual Exclusion is active

- Transfer

An observer cannot initiate a transfer while the observer is also observing a call.

If a user transfers a call, the observer is placed in the wait state. The observer is bridged onto the call when the transfer is complete.

Station Hunting

Use the Station Hunting feature to find an extension that is available to answer a call when the called extension is busy. The system checks for an idle extension in the station-hunting chain of the called extension before the system routes the call to the coverage path of the called extension.

The Station Hunting feature supports the following capabilities:

- **Station Hunting Before Coverage**
If the system finds an idle extension in the station-hunting chain of the called extension, the system leaves the call at the idle extension.
- **Station Hunting After Coverage**
If the system does not find an idle extension in the station-hunting chain of the called extension, the system routes the call to the coverage path of the last extension in the station-hunting chain.

Detailed description of Station Hunting

This section provides a detailed description of the Station Hunting feature.

To use Station Hunting, you create a station-hunting chain. This chain governs the order in which the system routes the calls when the system encounters a busy station. Each station in the chain links to only one subsequent station in the station-hunting chain. However, a station can be the receiving link from any number of station-hunting chains.

When the system starts to check the station-hunting chain, the system updates the display of the calling party with an *h*. The system also updates the display of the called party with an *h*.

You can administer an unlimited number of extensions in a station-hunting chain.

[Table 60, Routing calls through a Station Hunting chain](#), on page 1005 shows how the system routes a call through the station-hunting chain.

Table 60: Routing calls through a Station Hunting chain

Condition	Response
The extension is idle.	• The caller hears ringing. • The system does not continue to route the call through the station-hunting chain.
The extension is busy.	The system routes the call to the next extension in the station-hunting chain.
The hunt-to-station field on the Station screen of the extension is blank.	• The caller hears busy tone. • The system does not continue to route the call through the station-hunting chain.

1 of 2

Table 60: Routing calls through a Station Hunting chain

Condition	Response
The system encounters a station in the station-hunting chain for the second time.	• The caller hears busy tone. • The system does not continue to route the call through the station-hunting chain.
The system checks 30 extension in the station-hunting chain, and does not find an idle extension.	• The caller hears busy tone. • The system does not continue to route the call through the station-hunting chain.

2 of 2

Station Hunting and Call Coverage

You can administer the system to perform station hunting before the system sends calls to coverage, or after the system sends calls to coverage.

If you administer the system to use Station Hunting before the system uses Call Coverage, the system checks for a hunt-to station at the called extension. If the system finds a hunt-to station at the called extension, the system routes the call down the station-hunting chain. If the system does not find an idle extension in the station-hunting chain, the system routes the call to coverage.

Removing a station from a Station Hunting chain

When you remove a station from a station-hunting chain, the system attempts to maintain the chain. Consider the following examples:

- Station 1 links to station 2, and station 2 links to station 3. If you remove station 2, the system links station 1 to station 3.
- Station 1 links to station 2. Station 2 does not link to another extension. If you remove station 2, station 1 no longer links to another extension.

Duplicating a station in a Station Hunt chain

When you duplicate a station, the system does not copy the extension that is in the hunt-to station field on the **Station** screen to the station that you duplicate.

Hardware requirements for Station Hunting

The Station Hunting feature requires the following hardware:

- Telephone

Administering Station Hunting

The following steps are part of the administration process for the Station Hunting feature:

- [Assigning station hunting after coverage](#)
- [Assigning a hunt-to station to an extension](#)
- [Administering Station Hunting before Coverage](#)

This section describes:

- Any prerequisites for administering the Station Hunting feature
- The screens that you use to administer the Station Hunting feature
- Complete administration procedures for the Station Hunting feature

Prerequisites for administering Station Hunting

- None

Screens for administering Station Hunting

Screen name	Purpose	Fields
<i>Coverage Path</i>	Activate the Hunt after Coverage capability for a coverage path.	Hunt After Coverage
<i>Station</i>	Assign a hunt-to station to an extension.	Hunt-to Station
<i>System-Parameters Call Coverage/Call Forwarding</i>	Enable the Station Hunting before Coverage capability for your system.	Station Hunt Before Coverage

Assigning station hunting after coverage

To assign the Station Hunting after Coverage capability to a coverage path:

- 1 Type **change coverage-path *n***, where *n* is the number of the coverage path to which you want to assign the Station Hunting after Coverage capability. Press **Enter**.

The system displays the **Call Coverage** screen ([Figure 268, Coverage Path screen](#), on page 1008).

Figure 268: Coverage Path screen

change coverage path 2
Page 1 of 1

COVERAGE PATH

Coverage Path Number: 2

Next Path Number: ____ Hunt After Coverage: y

Linkage: ____

COVERAGE CRITERIA

Station/Group Status	Inside Call	Outside Call
Active?	n	n
Busy?	y	y
Don't Answer?	y	y
All?	n	n
DND/SAC/Goto Cover?	y	y

Number of Rings: 2

COVERAGE POINTS

Terminate to Coverage Pts. with Bridged Appearance? n

Point1: ____ Point2: ____ Point3: ____

Point4: ____ Point5: ____ Point6: ____

- 2 In the Hunt After Coverage field, perform one of the following actions:
 - Type **y** if you want the system to check the station-hunting chain of the last extension in the coverage path, if the system does not find an idle station in the coverage path.
 - Type **n** if you want the system to leave the call at the last available point in the coverage path.
- 3 Press **Enter** to save your changes.

Assigning a hunt-to station to an extension

To assign a hunt-to station to an extension:

- 1 Type **change station *n***, where ***n*** is the number of the extension to which you want to assign a hunt-to station. Press **Enter**.

The system displays the **Station** screen ([Figure 269, Station screen](#), on page 1009).

Figure 269: Station screen

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?	
Model:	Media Complex Ext:	
	IP Softphone? y	
	Remote Office Phone? y	

- 2 In the Hunt-to Station field, perform one of the following actions:
 - Type the extension that you want the system to check for an idle status, if the extension is busy.
 - Leave the field blank if you do not want the system to check further for an idle extension, if this extension is busy.
- 3 Press **Enter** to save your changes.

Administering Station Hunting before Coverage

To administer the Station Hunting before Coverage capability for your system:

- 1 Type **change system-parameters coverage-forwarding**. Press **Enter**.

The system displays the **System-Parameters Coverage-Forwarding** screen ([Figure 270, System Parameters-Call Coverage/Call Forwarding screen](#), on page 1010).

Figure 270: System Parameters-Call Coverage/Call Forwarding screen

```
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? _
```

- 2 In the **Station Hunt Before Coverage?** field, perform one of the following actions:
 - Type **y** to enable the Station Hunting Before Coverage capability for your system.
 - Type **n** to disable the Station Hunting Before Coverage capability for your system.
- 3 Press **Enter** to save your changes.

Reports for Station Hunting

The following reports provide information about the Station Hunting feature:

- The List Usage report shows all the extensions that use an extension as the hunt-to-station.

For detailed information on these reports and the associated commands, [click here](#), or see *Reports for Avaya Communication Manager*.

Considerations for Station Hunting

This section provides information about how the Station Hunting feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Station Hunting under all conditions. The following considerations apply to Station Hunting:

- None

Interactions for Station Hunting

This section provides information about how the Station Hunting feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Station Hunting in any feature configuration.

Remember that the system checks the station-hunting chain only for idle and available extensions.

- Adjunct Switch Applications Interface (ASAI)

The system checks the station-hunting chain when ASAI routes a call to an extension with a hunt-to station.

- Automatic Call Distribution (ACD)

An agent extension can be part of a station-hunting chain. The system checks the station-hunting chain of the agent only when the caller places the call directly to the agent extension. The system does not check a station-hunting chain for calls that the system routes through hunt groups to an ACD agent.

The system does not check a station-hunting chain for calls that are made to an extension for a logical agent.

- Automatic Callback

The system does not check the station-hunting chain of the called extension when the call is a callback-return call.

- Bridged Appearance

The system checks the station-hunting chain of an extension if the principal station does not have a call appearance at which the call can terminate. The system checks the chain, even though the extension has available bridged appearances on other stations.

- Busy Verification

The system does not check a station-hunting chain for busy-verify calls.

- Call Coverage

Call Coverage has precedence over Station Hunting.

The system uses Station Hunting for the last coverage point of the last station in the station-hunting chain under the following conditions:

- The **Hunt After Coverage** field, on the **Call Coverage** screen is set to **y**.
- The last coverage point is unavailable because the coverage point is busy, or no one answers the call.
- The last coverage point is a station that has an assigned hunt-to station.
- No one in the coverage path answered the call.

Coverage Don't Answer covers the call after the system checks the station-hunting chain, if the call can terminate at the coverage point, but no one answers the call.

If Station Hunting before Coverage is enabled on your system, the system checks the station-hunting chain of the called extension, before the system routes the call to the coverage path of the called extension.

The system routes the call to the coverage path of the called extension unless the called extension is associated with an XDID telephone. If the called extension is associated with an XDID telephone, the system routes the call to the coverage path of the non-XDID extension in the hunt-to field of the XDID station.

- **Call Detail Recording (CDR)**
CDR records the called extension, not the extension of the user who answers the call.
- **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, the system forwards the call. If the forwarded-to station is busy, the call follows the station-hunting chain of the forwarded-to extension.
- **Call Park**
The system does not check a station-hunting chain for callpark-return calls.
- **Call Pickup**
Station Hunting does not change the operation or the characteristics of Call Pickup.
- **Call Vectoring**
You cannot type a vector directory number (VDN) as a hunt-to station.
If the system encounters **with cov y** in a **route-to** command, the system routes a call to a busy station to the station-hunting chain of the station. The system does not route the call to the coverage path of the station. See the *Avaya Communication Manager Contact Center 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 checks the station-hunting chain. If the system cannot terminate the call to a station in the station-hunting chain, the call waits at the called extension.
- **Class of Restriction (COR)**
The system checks the COR of the called extension. The system does not check the COR of the stations in a station-hunting chain.
- **Distributed Communications System (DCS)**
Station Hunting is not a DCS feature. All stations in a station-hunting chain must be on the same server that runs Avaya Communication Manager.
- **Do Not Disturb**
If Do Not Disturb is active at a station, the system does not check a station-hunting chain for a call to the station.
- **Extension Number Portability (ENP)**
You cannot assign a remote ENP extension as a hunt-to station.
- **Hunting/Hunting Group**
You cannot assign a direct departmental calling (DDC) or uniform call distribution (UCD) extension as a hunt-to station.
- **Intercom Call**
The system denies Station Hunting for intercom calls to a busy extension.

- **Leave Word Calling (LWC)**
If a caller starts 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 the system checks a station-hunting chain for the call.
- **Night Service**
The system denies Station Hunting when a night service call is made to a busy night-console extension.
- **Outgoing Trunk Queueing (OTQ)**
The system does not attempt Station Hunting for an OTQ callback-return call.
- **Personal Central Office Line (PCOL)**
The system does not attempt Station Hunting for a PCOL call.
- **Personal Station Access (PSA)**
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. Note that ***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. Note, however, ***the system does not check tenant restrictions on hunt-to stations.***
- **Terminal Translation Initialization (TTI)**
The system considers a station with TTI separation as busy and bypasses the station in the station-hunting chain.
- **Terminating Extension Group (TEG)**
You cannot assign a TEG as a hunt-to station.
- **Uniform Dial Plan (UDP)**
You cannot assign a remote UDP extension as a hunt-to station.
- **X-port extension**
You can assign a hunt-to station to a station administered with **x** in the **Port** field of the **Station** screen. The system bypasses a hunt-to station with an **x** in the **Port** field of the **Station** screen.

Station Lock

Use the Station Lock feature to lock a telephone to prevent others from placing outgoing calls from the telephone.

Detailed description of Station Lock

This section provides a detailed description of the Station Lock feature.

With the Station Lock feature, users can lock the telephone to prevent others from placing outgoing calls from the telephone.

A user with an analog telephone uses a feature access code (FAC) to lock the telephone. A user with a digital telephone can use a feature access code (FAC) or a feature button to lock the telephone. Station Lock:

- Blocks unauthorized outgoing calls
- Allows outgoing emergency calls
- Allows incoming calls

The feature button lights when the user presses the button to activate Station Lock. Then, when a user attempts to place an outgoing call, the system generates a special dial tone to indicate that the Station Lock feature is active.

If a digital telephone has a feature button for Station Lock, but uses an FAC to activate the feature, the LED lights. The system does not generate the special tone.

If a digital telephone does not have a feature button for Station Lock, and uses an FAC to activate the feature, the system generates the special tone.

Avaya recommends that a user of a digital telephone use a Station Lock button, instead of an FAC, to activate Station Lock.

Any user who knows the system-wide FAC for Station Lock, and the Station Security Code (SSC) of a specific telephone, can lock or unlock the telephone.

A user can also lock or unlock a telephone from a remote location.

The attendant console can lock or unlock other telephones. The attendant console cannot be locked.

Hardware requirements for Station Lock

The Station Lock feature requires the following hardware:

- None

Administering Station Lock

This section provides the screens that you need to administer the Station Lock feature.

Screens for administering Station Lock

Screen name	Purpose	Fields
<i>COR</i>	Administer a Class of Service (COR) that allows the user to activate Station Lock with a feature access code (FAC).	Station Lock COR
<i>Feature Access Code (FAC)</i>	Assign a FAC for Station Lock.	Station Lock Activation
<i>Station</i>	Assign the user a COR that allows the user to activate Station Lock with an FAC.	COR
	Assign a Station Lock feature button for a user.	Feature Button • sta-lock
	Assign a Station Security Code (SSC) for a user.	Security Code

End-user procedures for Station Lock

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Station Lock:

- 1 Activating or deactivating Station Lock from a remote telephone

To activate or deactivate Station Lock from a remote telephone:

- 2 Dial a valid barrier code.
The system generates a dial tone.
- 3 Dial the feature access code (FAC) for Station Lock.
- 4 Dial the extension number.
- 5 Dial the Station Security Code (SSC).

Reports for Station Lock

The following reports provide information about the Station Lock feature:

- None

Considerations for Station Lock

This section provides information about how the Station Lock feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Station Lock under all conditions. The following considerations apply to Station Lock:

- None

Interactions for Station Lock

This section provides information about how the Station Lock feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Station Lock in any feature configuration.

- Attendant Console

You cannot lock an attendant consoles but you can lock a digital station that has console permissions.

You can dial the feature access code (FAC) for Station Lock from the attendant console in an attempt to remotely activate or deactivate Station Lock for another telephone.

- Personal Station Access (PSA)

You can use Station Lock to lock a PSA telephone, if the telephone has an extension. When a telephone is dissociated, you cannot activate Station Lock from the telephone.

Station Lock

Interactions for Station Lock

Station Security Code

Use the Station Security Code (SSC) to deny other users access to the functions associated with your station. Each station user can change their own SSC if they know the station's current settings.

Detailed description of Station Security Code

This section provides a detailed description of the Station Security Code (SCC) feature.

SCC enhances system security. To use SCC, you must create a system-wide SSC change FAC before users can change their SSC. You must also provide users with their individual SSC. A user cannot change a blank SSC.

Hardware requirements for Station Security Code

The Station Security Code feature requires the following hardware:

- None

Administering Station Security Code

The following steps are part of the administration process for the Station Security Code feature:

- [Creating a station security code](#)

This section describes:

- The screens that you use to administer the Station Security Code feature
- Complete administration procedures for the Station Security Code feature

Screens for administering Station Security Code

Screen name	Purpose	Fields
<i>Feature Access Code</i>	Set the access code for the feature.	Station Security Code Change Access Code
<i>Security-Related System Parameters</i>	Set the minimum length of the code.	Minimum Station Security Code Length
<i>Station</i>	Set the code for the station extension.	Security Code

Creating a station security code

To create a station security code:

- 1 Type **change feature-access-codes**. Press **Enter**.
The system displays the *Feature Access Code (FAC)* screen.
- 2 Type **#5** in the **Station Security Code Change Access Code** field. Press **Enter**.
This action sets the access codes for this feature. The Command prompt appears.
- 3 Type **change system-parameters security**. Press **Enter**.
The system displays the *Security-Related System Parameters* screen.
- 4 Type **7** in the **Minimum Station Security Code Length** field. Press **Enter**.
This action determines the minimum required length of the Station Security Codes that you enter on the *Station* screen. Longer codes are more secure. If you use station security codes for external access to telecommuting features, Avaya recommends that you use a minimum length of seven digits.
- 5 Type **change station n**, where **n** is the extension that you configured for telecommuting. Press **Enter**.
The system displays the *Station* screen ([Figure 271, Station screen](#), on page 1020).

Figure 271: Station screen

change station 1014		Page 1 of X
STATION		
Extension: 1014	Lock Messages? n	BCC: 0
Type:	Security Code: 7654321	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?	
Model:		
	Media Complex Ext:	
	IP Softphone? y	
	Remote Office Phone? y	

- 6 Type a number in the **Security Code** field that is equal in length to the number of digits that you entered in the *Security-Related System Parameters* screen.
In this example, type **7654321** in the **Security Code** field. This number is equal to the minimum number of digits that you entered in the *Security-Related System Parameters* screen.
- 7 Press **Enter** to save your changes.

Reports for Station Security Code

The following reports provide information about the Station Security Code feature:

- None

Considerations for Station Security Code

This section provides information about how the Station Security Code feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Station Security Code under all conditions. The following considerations apply to Station Security Code:

- None

Interactions for Station Security Code

This section provides information about how the Station Security Code feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Station Security Code in any feature configuration.

Users need a station security code to use the following system features and capabilities:

- Demand printing
- Extended User Administration of Redirected Calls
- Leave Word Calling
- Personal Station Access
- Voice Message Retrieval
- Station Lock
- Terminal Self-Administration

Supporting TTY Callers

Use the Supporting TTY Callers feature to enable callers to use a teletypewriter device (TTY) to listen to announcements that play TTY recordings. You can also use this feature to set up hunt groups for TTY callers, and to create vectors that process both TTY callers and voice callers.

TTY is also known as TDD (Telecommunications Device for the Deaf).

Detailed description of Supporting TTY Callers

This section provides a detailed description of the Supporting TTY Callers feature.

Reliable transmission of TTY information complies with the requirements and guidelines that are outlined in United States accessibility-related laws. Those laws include:

- Titles II, III, and IV of the Americans with Disabilities Act (ADA) of 1990.
- Sections 251 and 255 of the Telecommunications Act of 1996.
- Section 508 of the Workforce Investment Act of 1998.

Communication Manager TTY support is currently restricted to TTY devices that use either the:

- US English standard TTY protocol, specified by ANSI/TIA/EIA 825 as: “A 45.45 Baud FSK modem.”
- UK English standard TTY protocol, Baudot 50.

Important characteristics of the standards are:

- TTYs are silent when not transmitting. Unlike fax machines and computer modems, TTYs have no “handshake” procedure at the start of a call, nor do they have a carrier tone during the call. This approach has the advantage of permitting TTY tones, DTMF, and voice to be intermixed on the same call.
- The ability to intermix voice and typed TTY data on the same call. The most common usage is by people who are hard of hearing, but who can speak clearly. These people often prefer to receive text on a TTY device, and then speak in response. This process is referred to as Voice Carry Over (VCO).
- Operation is “half duplex.” TTY users must take turns transmitting and typically cannot interrupt each other. If two people try to type at the same time, two TTY devices might show no text at all or show text that is unrecognizable. Also, no automatic mechanism exists to let TTY users know when a character that the user correctly typed was received incorrectly.
- Each TTY character consists of a sequence of seven individual tones. The first tone is always a “start tone” at 1800 Hz. This tone is followed by a series of five tones, at either 1400 or 1800 Hz, which specify the character. The final tone in the sequence is always a “stop tone” at 1400 Hz. The stop tone is a border that separates this character from the next.

The following types of systems support TTY communication:

- Analog telephones and trunks
- Digital telephones and trunks

- VoIP gateways
- Messaging systems
- Automated attendant systems
- Interactive Voice Response (IVR) systems
- Wireless systems in which a TTY-compatible coder is used

Setting up announcements for TTY callers

TTY devices typically resemble a laptop computer. TTY devices have a one line-or a two-line alphanumeric display, instead of the computer screen.

You record announcements for TTY callers in the same way as you record voice announcements. However, instead of recording from the handset of your telephone, you record from a TTY device. The device is attached to your telephone. You use an acoustic coupler into which you place the telephone handset or by plugging the TTY device directly into the back, if it is a digital phone. After calling the announcement extension, and pressing **1** to record. To use the device, you type the announcement using the TTY device.

If you use an acoustic coupler to connect your telephone for recording, you can record TTY and voice into a single announcement. In this case, when you press **1** to record, you can type the TTY message, then immediately pick up the handset to record the voice message. For this type of recording, digital phones also offer the option to press **#** to complete the recording, which eliminates any extraneous noise at the end of the recording. Unfortunately, using this method for combined TTY and voice recordings is likely to create extraneous noise in the middle of your announcements.

As a alternative to recording with your telephone, you can create .WAV files on other recording applications and you can then copy and save the .WAV files to your announcement board.

Setting up hunt groups for TTY callers

In a call center, TTY callers can be accommodated by a hunt group that includes TTY-equipped agents. Although many TTYs can connect directly with the telephone network by way of analog RJ-11 jacks, Avaya recommends that agents be equipped with TTYs that include an acoustic coupler that can accommodate a standard telephone handset. One reason for this recommendation is that a large proportion of TTY users are hearing impaired, but can speak clearly. These individuals often prefer to receive calls on a TTY and then speak in response. This requires the call center agent to alternate between listening on the telephone and then typing on the TTY. An acoustically coupled configuration makes this process considerably easier.

Although TTY-emulation software packages are available for personal computers, most of these packages do not have the ability to intermix voice and TTY on the same call.

For a TTY hunt group, you can record TTY announcements and use them for the hunt group queue. To record announcements for TTY, follow the same steps as with voice recordings from your telephone. However, instead of speaking into your phone to record, you type the announcement with the TTY device.

For an alternative to creating a TTY hunt group, you can use vectors to process TTY calls. With vectors, you can allow TTY callers and voice callers to use the same phone number. In this case, you can also record a single announcement that contains both TTY signaling and a voice recording.

Handling TTY calls with vectors

Unlike fax machines and computer modems, a Tele-typewriter device (TTY) has no handshake tone and no carrier tone. A TTY is silent when not transmitting. This is why systems cannot identify TTY callers automatically. However, the absence of these special tones also means that voice and TTY tones can be intermixed in prerecorded announcements. The ability to provide a hybrid voice-and-TTY announcement, when combined with the automated attendant vectoring capability, can permit a single phone number to accommodate both voice and TTY callers.

The following sample shows a vector that allows TTY callers to access a TTY agent. It begins with a step that plays a TTY announcement combined with a voice announcement. The announcement tells the TTY caller to dial a digit that will direct them to a TTY support person. The vector then processes the digit entered to connect the TTY caller to the TTY split (or hunt group).

In the following example, split 47 (hunt group 47) has already been established and consists of TTY-enabled agents.

If a TTY caller calls the number that connects to vector 33, the following actions occur:

- 1** After a short burst of ringing, a quick burst of TTY tones is sent to the caller to tell the caller to hold, HD. Then a voice announcement is played for callers using a normal telephone connection. The announcement tells them to stay on the line. Finally, another burst of TTY tones is sent to the TTY caller which displays on the caller's TTY device as, "Dial 1."

The TTY caller does not hear the voice announcement, but because the step collects digits, the steps allows the caller to dial a **1** from a touchtone telephone.

NOTE:

For voice callers, the burst of TTY tones lasts about one second and sounds like a bird chirping.

- 2** In vector step 3, since the TTY caller entered **1** in vector step 2, the TTY caller is sent to vector step 8. At this point, the caller is put in queue for a TTY-enabled agent in split 47.

NOTE:

The voice caller is sent to vector step 3 also, but a voice caller does not go to vector step 8 because the caller did not enter **1** at vector step 2. Instead, voice callers continue on to vector step 4, where they connect to split 48.

- 3** While the TTY caller waits in queue, the caller hears silence from vector step 9, and then the announcement in vector step 10, and is then looped back to wait with silence by vector step 11.

Hardware requirements for Supporting TTY Callers

The Supporting TTY Callers feature requires the following hardware:

- None

Administering Supporting TTY Callers

This section describes the screens that you use to administer the Supporting TTY Callers feature.

Screens for administering Supporting TTY Callers

- None

Reports for Supporting TTY Callers

The following reports provide information about the Supporting TTY Callers feature:

- None

Considerations for Supporting TTY Callers

This section provides information about how the Supporting TTY Callers feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Supporting TTY Callers under all conditions. The following considerations apply to Supporting TTY Callers:

- None

Interactions for Supporting TTY Callers

This section provides information about how the Supporting TTY Callers feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Supporting TTY Callers in any feature configuration.

- None

Telephone Display

Use the Telephone Display feature to provide users of multiappearance telephone with current call and message information. The information that appears depends on the type of display that the user selects with the buttons on the telephone.

Using the features of the telephone, users can retrieve stored information, such as messages and directory information. You can select English, French, Italian, Spanish, user-defined, or Unicode languages to display the messages or the information.

Telephone Display supports the following capabilities:

- Button display modes
- Call-related information display
- Message retrieval
- Feature information displays
- Enhanced Telephone Display

Detailed description of Telephone Display

This section provides a detailed description of the Telephone Display feature.

Use the Telephone Display feature to provide users of multiappearance telephones with current call and message information. The information that appears depends on the type of display that the user selects with the buttons on the telephone.

Using the features of the telephone, users can retrieve stored information, such as messages and directory information. You can select English, French, Italian, Spanish, user-defined, or Unicode languages to display the messages or information.

With [Enhanced Telephone Display](#), you can choose the types of characters that appear on the telephone display of the user. You can administer the software to display Roman (European) characters, or Cyrillic (Russian), Katakana (Japanese), or Ukrainian characters. The telephones that your company uses determine the character sets that you can display.

Button display modes

You can assign several display modes to telephone buttons. To access these modes, users press the assigned button on the telephone. All the buttons are administrable.

Button mode	Function
Normal	Displays call-related information for the active call appearance. This information includes call appearance, and the name and the number of the calling- or called-party, depending on the type of call. Can also display elapsed time when the display is in normal mode. The system shows the elapsed time in hours, minutes, and seconds. Timing starts and stops when the button is pressed.
Inspect	Displays call-related information for an incoming call when the user is active on a different call appearance. Users 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 that is stored in an Abbreviated Dialing button that is administered to the telephone • A number stored in an Abbreviated Dialing list • A number assigned to a button that is administered by Facility Busy Indication
Date and Time	The current date and time of day.
Integrated Directory	Turns off the touchtone signals and allows the user to use the touchtone buttons to enter the name of a system user. After the user enters a name, the display shows the name and the extension. Integrated Directory can use one additional button: • Call-Disp automatically returns the call that is requested by the currently displayed message or the currently displayed name and extension.
Message Retrieval	Retrieves messages for telephone users. If no messages are stored, the display shows NO MESSAGES. Users can retrieve messages 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 that is requested by the currently displayed message or the currently displayed name and extension.

Button mode	Function
Coverage Message Retrieval Mode	Retrieves messages for telephone users who do not have a display module assigned to their phone. You must administer retrieval permission for a user to retrieve messages of other users. The user does not need to lift the handset to retrieve messages. The user can retrieve messages even if the retriever is active on a call. Coverage Message Retrieval can use three 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.

Call-related information display

The software provides the following call-related information:

- Call appearance identification

A lowercase letter is used to designate call appearance buttons on the display. The display shows a= for an incoming call on the first button, b= for an incoming call g on the second button, and so on.

The system might 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 name of the caller or a unique identification that is administered for the telephone being used, along with the extension of the calling party. When the call is from outside the system, the display shows the trunk group name (such as CHICAGO) and the Trunk Access Code (TAC) that is assigned to the trunk group used for the call. If a user is active on a call and receives another call, the display automatically shows the identification of the second caller for a few seconds. The system 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:

Because of space limitations, some name displays are shortened to 15 characters. These include displays for transferred or covered calls, non-DCS, ISDN-PRI calls, VDN service observing displays, LWC messages, or the queue status of an agent.

- **Called party identification**

On calls to a system user, the digits appear on the display as the digits are dialed. After dialing is complete, the name and the extension of the called party appears. If no name is accessed, the dialed digits remain on the display.

On outgoing calls, the digits appear on the display as the digits are dialed. After dialing is complete, the display shows the name and the TAC that is 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 the TAC.

For example:

Dialed digits

a=3602

then

a=TOM BROWN

3062

or, if no name is available

a=EXT 3602

3062

- **Call purpose**

Call purpose identifies the reason for an incoming call or a redirected call. The system does not identify a call purpose for a normal incoming call. 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 telephone unanswered. This message also indicates that the called user has a temporary bridged appearance of the call.
f — (Call Forwarding)	Another user has forwarded calls to this telephone.
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 the call of a Call Pickup group member.

Display	Meaning
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 was redirected to this phone.

Message retrieval

You can designate certain phones and attendant groups for system-wide message retrieval. Users of these telephones or consoles can retrieve Leave Word Calling (LWC) and call coverage messages for other telephone users. These other users can include Direct Department Calling (DDC) groups, Uniform Call Distribution (UCD) groups, and Terminating Extension Groups (TEG). Users of these telephones or consoles can also retrieve external call logs. You can assign system-wide retrieving telephones or consoles. Use the **Feature-Related System Parameters** screen.

Messages for a telephone user can be retrieved at selected telephones, or any attendant console. However, the retriever must be on the call coverage path of the user, and permission to retrieve messages must be assigned for the telephone of the user.

Feature information displays

Telephone displays provide information about the activity on individual telephones and consoles, including confirmation that a certain feature is being used. You administer the language used for messages on the **Station** screen for each telephone.

Enhanced Telephone Display

With Enhanced Telephone Display, you can choose the types of characters that appear on your telephone 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 also depends on the telephones that your company uses.

You can choose one of the following character sets for messages on your display telephones:

- Cyrillic contains the characters that are required to display the Russian language. All Russian characters appear in capital letters.
- Katakana contains the characters that are required to display the Japanese language, and as some European characters and other symbols. All Japanese characters appear in uppercase letters.
- Roman contains two character sets:
 - US English contains the Roman alphabet, and the numerals, and special characters on the standard US English keyboard. US English characters appear in uppercase and lowercase letters.
 - European contains characters for many European languages. All European characters appear in capital letters.

- Ukrainian contains the characters that are required to display the Ukrainian language. All Ukrainian characters appear in uppercase letters.

The type of telephones that your company uses must support the characters that you want to display. Each character set requires specific firmware in the telephone. Ensure that you use telephones with the same firmware type across your entire system. If you do not use telephones with the same firmware type across your entire system, the displays do not appear as expected. Your Avaya representative can ensure that you have the correct telephone types for the characters that you want to display.

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 61: Automatic Wakeup

English	French	Italian	Spanish
<i>AUTO WAKEUP - Ext: xxxxx Time: -- :-- xM</i>	<i>REVEIL AUTO. - POSTE: xxxxx HEURE: --:--</i>	<i>SERVIZIO SVEGLIA - Tel: xxxxx Ora: --:--</i>	<i>DESPERT AUTOMA - EXT: xxxxx HORA: --:--</i>
<i>INVALID EXTENSION - TRY AGAIN</i>	<i>NUMERO DE POSTE EST ERRONE - REESSAYER</i>	<i>NUMERO ERRATO - RIPETERE</i>	<i>EXTENSION NO VALIDO - INTENTE DE NUEVO</i>
<i>WAKEUP ENTRY DENIED - INTERVAL FULL</i>	<i>DEM. REVEIL REFUSEE - INTERVALLE PLEIN</i>	<i>SVEGLIA NON ATTIVATA - ORARIO OCCUP</i>	<i>ENTRADA DENEGADA - INTERVALO COMPLETO</i>
<i>WAKEUP ENTRY DENIED - NO PERMISSION</i>	<i>DEM. REVEIL REFUSEE - SANS AUTORISATION</i>	<i>SVEGLIA NON ATTIVATA - NON PERMESSO</i>	<i>ENTRADA DENEGADA - SIN PERMISO</i>
<i>WAKEUP ENTRY DENIED - SYSTEM FULL</i>	<i>DEM. REVEIL REFUSEE - ENCOMBREMENT</i>	<i>SVEGLIA NON ATTIVATA - CONGESTIONE</i>	<i>ENTRADA DENEGADA - SISTEMA COMPLETO</i>
<i>WAKEUP ENTRY DENIED - TOO SOON</i>	<i>DEM. REVEIL REFUSEE - TROP TOT</i>	<i>SVEGLIA NON ATTIVATA - TROPPO PRESTO</i>	<i>ENTRADA DENEGADA - MUY PRONTO</i>
<i>WAKEUP REQUEST CANCELED</i>	<i>DEMANDE DE REVEIL EST ANNULEE</i>	<i>RICHIESTA SVEGLIA CANENTRYATA</i>	<i>SOLICITUD DE DESPERTADOR CANCELADA</i>
<i>WAKEUP REQUEST CONFIRMED</i>	<i>DEMANDE DE REVEIL EST CONFIRMEE</i>	<i>RICHIESTA SVEGLIA CONFERMATA</i>	<i>SOLICITUD DE DESPERTADOR CONFIRMADA</i>
<i>Wakeup Call</i>	<i>APPEL DE REVEIL</i>	<i>Serv. Sveglia</i>	<i>Despierte</i>

Table 62: ASAI

English	French	Italian	Spanish
<i>You have adjunct messages</i>	<i>MESSAGES SUPPLEMENTAIRES</i>	<i>MESSAGGI AGGIUNTIVI</i>	<i>TIENE MENSAJES ADICIONALES</i>

Table 63: Busy verification of terminals and trunks

English	French	Italian	Spanish
<i>ALL MADE BUSY</i>	<i>TOUS OCC.</i>	<i>TUTTI OCCUPATI</i>	<i>TODAS OCUPADAS</i>
<i>BRIDGED</i>	<i>EN DERIVATION</i>	<i>OCCUPATO</i>	<i>PUENTEADA</i>
<i>DENIED</i>	<i>INTERDIT</i>	<i>NON PERMESSO</i>	<i>DENEGADO</i>
<i>INVALID</i>	<i>ERRONE</i>	<i>NON VALIDO</i>	<i>NO VALIDO</i>
<i>NO MEMBER</i>	<i>AUCUN MEMBRE</i>	<i>NESSUN ELEMENTO</i>	<i>NINGUN MIEMBRO</i>
<i>OUT OF SERVICE</i>	<i>HORS SERVICE</i>	<i>FUORI SERVIZIO</i>	<i>FUERA SERVICIO</i>
<i>RESTRICTED</i>	<i>RESTREINT</i>	<i>RISTRETTO</i>	<i>RESTRINGIDO</i>
<i>TERMINATED</i>	<i>TERMINE</i>	<i>TERMINATO</i>	<i>TERMINADO</i>
<i>TRUNK SEIZED</i>	<i>CIRCUIT SAISI</i>	<i>GIUNZIONE IMP.</i>	<i>ENLACE OCUPADO</i>
<i>VERIFIED</i>	<i>VERIFIE</i>	<i>VERIFICATO</i>	<i>VERIFICADO</i>

Table 64: 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 65: Call Detail Recording

English	French	Italian	Spanish
<i>CDR OVERLOAD"</i>	<i>SURCHARGE EDA</i>	<i>SVRACCARICO DAC</i>	<i>SOBRECARG A DAT</i>

Table 66: Call progress feedback displays

English	French	Italian	Spanish
<i>busy</i> (Extension Busy, Intrusion Not Allowed, Call Waiting Not Allowed)	<i>OCCUPE</i> (Occupe)	<i>occ</i> (Occupato)	<i>OCUPADA</i> (Ocupada)
<i>busy(I)</i> (Extension Busy, Intrusion Allowed, Call Waiting Not Allowed)	<i>OCC.(E)</i> (Entree ligne occupe)	<i>occ(I)</i> (Occupato- Intrusione)	<i>OCUP(I)</i> (Ocupada- intrusion)
<i>ringing</i> (Extension Ringing)	<i>SONNE</i> (Libre)	<i>libero</i> (Libero)	<i>LIBRE</i> (Libero)
<i>wait</i> (Extension Busy, Intrusion Not Allowed, Call Waiting Allowed)	<i>ATTENTE</i> (Attente)	<i>auat</i> (Autoattesa)	<i>ESPERA</i> (Espera)
<i>(I) wait</i> (Extension Busy, Intrusion Allowed, Call Waiting Allowed)	<i>(E) ATTENTE</i> (Entree ligne attente)	<i>(I) auat</i> (Intrusione- Autoattesa)	<i>(I) ESPERA</i> (Intrusion, en espera)

Table 67: 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.

Figure 272: Date/Time Mode and Formats – English

<DATE/TIME>	<TIME><DATE>
<TIME>	<HR>:<MIN><M>
<HR>	1-12 (hour of day, no leading zeroes)
<MIN>	00-59 (minute of hour)
<M>	"am" or "pm"
<DATE>	<DOW><MONTH><DOM>, <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
	Blank

Figure 273: Date/Time Mode and Formats – French, Italian, Spanish, and User-Defined

<DATE/TIME>	<TIME><DATE>
<TIME>	<HR>:<MIN>
<HR>	0-23 (hour of day, no leading zeroes)
<MIN>	00-59 (minute of hour)
<DATE>	<DOW><DOM><MONTH>, <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
	Blank

Table 68: 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 69: Date/Time mode — time not available

<i>English</i>	<i>French</i>	<i>Italian</i>	<i>Spanish</i>
<i>SORRY, TIME UNAVAILABLE NOW</i>	<i>HEURE ET DATE INDISPONIBLES</i>	<i>ORA E DATA TEMPO DISPONIBILI</i>	<i>HORA Y FECHA NO DISPONIBLE S AHORA</i>

Table 70: 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 71: Do Not Disturb (Hotel/Motel feature)

<i>English</i>	<i>French</i>	<i>Italian</i>	<i>Spanish</i>
<i>DO NOT DIST - Group: xx Time: -- :-- xM</i>	<i>NE PAS DERANGER GROUPE: xx HEURE: -- :--</i>	<i>NON DISTURBARE - Grp: xx Ora: --:--</i>	<i>NO MOLESTAR - GRUPO: xx HORA: --:--</i>
<i>DO NOT DIST - Ext: xxxxx Time: -- :-- xM</i>	<i>NE PAS DERANGER POSTE:xxxxx HEURE: -- :--</i>	<i>NON DISTURBARE - Tel: xxxxx Ora: --:--</i>	<i>NO MOLESTAR - EXT: xxxxx HORA: --:--</i>
<i>DO NOT DIST ENTRY DENIED - INTERVAL FULL</i>	<i>DEMANDE EST REFUSEE - INTERVALLE PLEIN</i>	<i>SERVIZIO NON ATTIVATO - ORARIO OCCUP</i>	<i>ENTRADA DENEGADA - INTERVALO COMPLETO</i>

1 of 2

Table 71: Do Not Disturb (Hotel/Motel feature)

<i>English</i>	<i>French</i>	<i>Italian</i>	<i>Spanish</i>
<i>DO NOT DIST ENTRY DENIED - NO PERMISSION</i>	<i>DEMANDE EST REFUSEE - SANS AUTORISATION</i>	<i>SERVIZIO NON ATTIVATO - NON PERMESSO</i>	<i>ENTRADA DENEGADA - SIN PERMISO</i>
<i>DO NOT DIST ENTRY DENIED - SYSTEM FULL</i>	<i>DEMANDE EST REFUSEE - ENCOMBREMENT</i>	<i>SERVIZIO NON ATTIVATO - CONGESTIONE</i>	<i>ENTRADA DENEGADA - SISTEMA COMPLETO</i>
<i>DO NOT DIST ENTRY DENIED - TOO SOON</i>	<i>DEMANDE EST REFUSEE - TROP TOT</i>	<i>SERVIZIO NON ATTIVATO - TROPPO PRESTO</i>	<i>ENTRADA DENEGADA - MUY PRONTO</i>
<i>INVALID GROUP - TRY AGAIN</i>	<i>GROUPE ERRONE - REESSAYER</i>	<i>GRUPPO NON VALIDO - RIPETERE</i>	<i>GRUPO NO VALIDO - INTENTE DE NUEVO</i>
<i>THANK YOU - DO NOT DIST ENTRY CONFIRMED</i>	<i>MERCI - DEMANDE EST CONFIRMEE</i>	<i>NON DISTURBARE - RICHIESTA CONFERMATA</i>	<i>NO MOLESTAR - ENTRADA CONFIRMADA</i>
<i>THANK YOU - DO NOT DIST REQUEST CANCELED</i>	<i>MERCI - DEMANDE EST ANNULEE</i>	<i>NON DISTURBARE - RICHIESTA CANENTRYATA</i>	<i>MUCHAS GRACIAS - SOLICITUD CANCELADA</i>
2 of 2			

Enhanced Abbreviated Dialing -user defined

Figure 274: Language Translations screen

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 72: 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 73: Integrated directory display model

English	French	Italian	Spanish
<i>DIRECTORY - PLEASE ENTER NAME</i>	<i>ANNUAIRE - ENTRER LE NOM</i>	<i>ELENCO UTENTI - INTRODURRE NOME</i>	<i>GUIA TELEFONICA - INTRODUZCA NOMBRE</i>
<i>DIRECTORY UNAVAILABLE - TRY LATER</i>	<i>ANNUAIRE INDISPONIBLE - REESSAYER</i>	<i>ELENCO UTENTI TEMP. NON DISPONIBILE</i>	<i>GUIA TEL INDISPONIBLE - INTENTE DESPUES</i>
<i>NO MATCH - TRY AGAIN</i>	<i>INTROUVABLE - REESSAYER</i>	<i>NESSUNA CORRISPONDENZ A - RIPETERE</i>	<i>NO CORRESPONDE - INTENTE DE NUEVO</i>

Table 74: ISDN

English	French	Italian	Spanish
<i>ANSWERED BY</i>	<i>REPONDU PAR</i>	<i>RISPOSTA DA</i>	<i>RESPONDIDO POR</i>
<i>CALL FROM</i>	<i>APPEL DE</i>	<i>CHIAMATA DA</i>	<i>LLAMADA DE</i>
<i>INTL</i>	<i>INTL</i>	<i>INTL</i>	<i>INTL</i>

Leave Word Calling

Figure 275: Leave Word Calling Format – English

<CALLER_ID><DATE><TIME><M><C>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
	blank

Figure 276: Leave Word Calling Formats – French, Italian, Spanish, and User-Defined

<CALLER_ID><DATE><TIME><C>APPL<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

Table 75: Leave Word Calling messages

English	French	Italian	Spanish
<i>CANNOT BE DELETED - CALL MESSAGE CENTER</i>	<i>NE PEUT ETRE SUPP./APPELER RECEP. MESS.</i>	<i>NON CANENTRYATO. CHIAMARE CENTRO MESSAGGI</i>	<i>NO ELIMINADO-LLAMA CENTRO DE MENSAJES</i>
<i>DELETED</i>	<i>SUPPRIME</i>	<i>MESSAGGIO CANENTRYATO</i>	<i>ELIMINADO</i>
<i>END OF MESSAGES (NEXT TO REPEAT)</i>	<i>FIN DES MESSAGES (SUIVANT POUR REPETER)</i>	<i>FINE MESSAGGI. <successivo> PER RIPETERE</i>	<i>FIN DE MENSAJES (SIGUIENTE PARA REPITIR)</i>
<i>GET DIAL TONE, PUSH Cover Msg Retrieval</i>	<i>TONALITE D'ENVOI - <LECT. MESS. COUV.></i>	<i><rec mess copert> DOPO IL TONO DI CENTR</i>	<i>OBTENGA TONO OPRIMA <RECUP MNSJE COBERT></i>
<i>IN PROGRESS</i>	<i>EN COURS</i>	<i>ATTENDERE...</i>	<i>EN CURSO</i>
<i>MESSAGE RETRIEVAL DENIED</i>	<i>LECTURE DE MESSAGES INTERDITE</i>	<i>LETTURA MESSAGGIO NON PERMESSA</i>	<i>RECUPERACION DE MENSAJES DENEGADA</i>
<i>MESSAGE RETRIEVAL LOCKED</i>	<i>LECTURE DE MESSAGES BLOQUEE</i>	<i>LETTURA MESSAGGIO BLOCCATA</i>	<i>RECUPERACION DE MENSAJES BLOQUEADA</i>
<i>MESSAGES FOR</i>	<i>MESSAGES POUR</i>	<i>MESSAGGI PER</i>	<i>MENSAJES PARA</i>
<i>MESSAGES UNAVAILABLE - TRY LATER</i>	<i>MESSAGES INDISPONIBLES - REESSAYER</i>	<i>MESSAGGI TEMPORANEAMENTE NON DISPONIBILI</i>	<i>MENSAJES NO DISPONIBLES, INTENTE DESPUES</i>
<i>Message Center (AUDIX) CALL</i>	<i>APPEL DE LA RECEPTION DE MESS. (AUDIX)</i>	<i>Chiamata dal Centro Messaggi (AUDIX)</i>	<i>LLAMADA DEL CENTRO DE MENSAJES (AUDIX)</i>
<i>NO MESSAGES</i>	<i>PAS DE MESSAGES</i>	<i>NESSUN MESSAGGIO</i>	<i>NINGUN MENSAJE</i>
<i>WHOSE MESSAGES? (DIAL EXTENSION NUMBER)</i>	<i>MESSAGES DE QUEL NO.? (ENTRER NO. POSTE)</i>	<i>LETTURA MESSAGGI. INTRODURRE NUMERO TEL.</i>	<i>MENSAJES DE QUIEN? (MARCAR EXTENSION)</i>

Table 76: Malicious Call Trace

English	French	Italian	Spanish
<i>MALICIOUS CALL TRACE REQUEST</i>	<i>DEPISTAGE D'APPELS MALVEILLANTS</i>	<i>RICHIESTA RINTRACCIO CHIAMATE MALEVOLE</i>	<i>RASTREO DE LLAMADA MALINTENCION ADA</i>
<i>MCT activated by: for:</i>	<i>DAM ACTIVE par: pour:</i>	<i>RCM attivato da: per:</i>	<i>RLM activada por: para:</i>
<i>original call redirected from:</i>	<i>redirection appel initial de: (EXTENSION)</i>	<i>chiamata iniziale rinvia da:</i>	<i>llamada orig. transferida de:</i>
<i>party: (EXTENSION)</i>	<i>demandeur: (EXTENSION)</i>	<i>utente: (INTERNO)</i>	<i>usuario: (EXTENSION)</i>
<i>party: (ISDN SID/CNI)</i>	<i>demandeur: (NIP/INA ISDN)</i>	<i>utente: (NIC/INC ISDN)</i>	<i>usuario: (ISDN NIE/INU)</i>
<i>party: (PORT ID)</i>	<i>demandeur: (REF. PORT ISDN)</i>	<i>utente: (ID DELLA PORTA ISDN)</i>	<i>usuario: (ID DEL PUERTO ISDN)</i>
<i>party: (ISDN PORT ID)</i>	<i>demandeur: (REF. PORT)</i>	<i>utente: (ID DELLA PORTA)</i>	<i>usuario: (ID DEL PUERTO)</i>
<i>END OF TRACE INFORMATION</i>	<i>FIN DES INFO DE DEPISTAGE</i>	<i>INFORMAZIONI FINALI SUL RINTRACCIO</i>	<i>FIN DE INFORMACION DE RASTREO</i>
<i>voice recorder port:</i>	<i>port enregistreur vocal:</i>	<i>porta del registratore:</i>	<i>puerto de grabado de voz:</i>

Table 77: Caller information

English	French	Italian	Spanish
<i>Info:</i>	<i>INFO.:</i>	<i>Info:</i>	<i>INFORM:</i>

Table 78: Emergency access to attendant

English	French	Italian	Spanish
<i>a=xxxxxxxxxxxxxxxx Ext xxxxx xx in EMRG Q</i>	<i>a=xxxxxxxxxxxxxxxx x POSTE xxxxx xx FIL URG</i>	<i>a=xxxxxxxxxxxxxxxx Der xxxxx xx in C EMRG</i>	<i>a=xxxxxxxxxx xxxxx EXT xxxxx xx EN C EMRG</i>

Table 79: Queue status

English	French	Italian	Spanish
<i>HUNT GROUP</i> <i><x> NOT</i> <i>ADMINISTERED</i>	<i>GROUPE DE</i> <i>DIST. <x> NON</i> <i>ADMINISTRE</i>	<i>GRUPPO <x></i> <i>NON</i> <i>AMMINISTRATO</i>	<i>GRUPO</i> <i>BUSQUEDA <x></i> <i>NO</i> <i>ADMINISTRADO</i>

Table 80: Queue status indication

English	French	Italian	Spanish
<i><15 chrs> Q-time</i> <i>xx:xx calls xx</i>	<i><15 chrs></i> <i>TEMPS-F xx:xx</i> <i>APPELS xx</i>	<i><15 chrs> T-coda</i> <i>xx:xx chiam xx</i>	<i><15 chrs></i> <i>HORA-Cxx:xx</i> <i>LLAMADAS</i> <i>xx</i>

Table 81: Miscellaneous call identifier

English	French	Italian	Spanish
<i>sa</i> (ACD Supervisor Assistance)	<i>AS</i> (Assistance surveillant)	<i>as</i> (Assistenza Supervisoree)	<i>AS</i> (Ayuda de supervisor)
<i>ac</i> (Attd Assistance Call)	<i>AA</i> (Appel assistance)	<i>ao</i> (Assistenza Operatore)	<i>AO</i> (Ayuda de operadora)
<i>tc</i> (Attd Control Of A Trunk Group)	<i>CF</i> (Commande faisceau)	<i>fc</i> (Fascio Controllato)	<i>CE</i> (Control enlaces)
<i>an</i> (Attd No Answer)	<i>TR</i> (Telephoniste sans reponse)	<i>on</i> (Operatore Non Risponde)	<i>ON</i> (Operadora no responde)
<i>pc</i> (Attd Personal Call)	<i>AP</i> (Appel personnel)	<i>cp</i> (Chiamata Personale)	<i>LP</i> (Llamada personal)
<i>rc</i> (Attd Recall Call)	<i>RA</i> (Rappel)	<i>rc</i> (Richiamata)	<i>RL</i> (Rellamada)
<i>rt</i> (Attd Return Call)	<i>RE</i> (Retour)	<i>rt</i> (Ritornata)	<i>RT</i> (Retorno)
<i>sc</i> (Attd Serial Call)	<i>AS</i> (Appel en serie)	<i>ic</i> (Inoltro a Catena)	<i>LS</i> (Llamada en serie)

1 of 4

Table 81: Miscellaneous call identifier

English	French	Italian	Spanish
<i>co</i> (Controlled Outward Restriction)	<i>RD</i> (Restriction de depart)	<i>cu</i> (Controllata Uscente)	<i>RS</i> (Restriccion saliente)
<i>cs</i> (Controlled Station to Station Restriction)	<i>RP</i> (Restriction vers postes)	<i>cd</i> (Controllata Derivati)	<i>CS</i> (Control estacion)
<i>ct</i> (Controlled Termination Restriction)	<i>AR</i> (Restriction d'arrivee)	<i>ct</i> (Controllata Terminante)	<i>RE</i> (Restriccion entrante)
<i>db</i> (DID Find Busy Station With CO Tones)	<i>OP</i> (Occupation du poste)	<i>po</i> (Passante Occupata)	<i>EO</i> (Estacion occupada)
<i>da</i> (DID Recall Go To Attd)	<i>RT</i> (Rappel telephoniste)	<i>pr</i> (Richiamata su Passante)	<i>RD</i> (Rellamada directa)
<i>qf</i> (Emerg. Queue Full Redirection)	<i>FP</i> (File d'urgence pleine deviation)	<i>de</i> (Deviata Emergenza)	<i>DE</i> (Desvio de emergencia)
<i>hc</i> (Held Call Timed Reminder)	<i>AG</i> (Indicatif d'appel en garde)	<i>at</i> (Avviso Chiamata in tenuta)	<i>LR</i> (Recordatorio de llamada retenida)
<i>ic</i> (Intercept)	<i>IN</i> (Interception)	<i>in</i> (Intercettata)	<i>IN</i> (Intercepcion)
<i>ip</i> (Interposition Call)	<i>AI</i> (Appel interposition)	<i>ip</i> (Interposizione)	<i>EP</i> (Entre posiciones)
<i>ld</i> (LDN Calls on DID Trunks)	<i>SD</i> (Selection directe)	<i>pd</i> (Diretta Passante)	<i>LD</i> (Larga distancia)
<i>so</i> (Service Observing)	<i>ES</i> (ecoute du service)	<i>is</i> (Inclusione Supervisore)	<i>SS</i> (Supervision del servicio)
<i>na</i> (Unanswered or Incomplete DID Call)	<i>SR</i> (Sans reponse)	<i>pn</i> (Passante Non Risposta)	<i>SR</i> (Sin respuesta)
<i>ACB</i> (Automatic Callback)	<i>R. AUTO.</i> (Rappel automatique)	<i>PRN</i> (Prenotazione Automatica)	<i>RA</i> (Rellamada automatica)

2 of 4

Table 81: Miscellaneous call identifier

English	French	Italian	Spanish
<i>callback</i> (Callback Call)	<i>RAPPEL</i> (Rappel)	<i>prenotaz</i> (Prenotazione)	<i>RELLAM</i> (Rellamada)
<i>park</i> (Call Park)	<i>G. I.</i> (garde par) indicatif	<i>parch.</i> (Parcheggiata)	<i>ESTAC</i> (Estacionamiento de llamada)
<i>control</i> (Control)	<i>CONTROLE</i> (Controle)	<i>cntr.op.</i> (Controllo Operatore)	<i>CONTROL</i> (Control)
<i>ICOM</i> (Intercom Call)	<i>INTERCOM</i> (Intercommunicati on)	<i>ICOM</i> (Intercom)	<i>INTERF</i> (Llamda interfono)
<i>OTQ</i> (Outgoing Trunk Queuing)	<i>FFD</i> (File faisceaux de depart)	<i>RFO</i> (Richiamata su Fascio Occupato)	<i>EES</i> (Espera de enlace de salida)
<i>priority</i> (Priority Call)	<i>PRIORITE</i> (Appel prioritaire)	<i>priorita</i> (Priorita')	<i>PRIORIT</i> (Llamada prioritaria)
<i>recall</i> (Recall Call)	<i>APP.RAP.</i> (Appel rappel)	<i>richiam</i> (Richiamata)	<i>REPET</i> (Rellamada)
<i>return</i> (Return Call)	<i>RETOUR</i> (Retour)	<i>ritorno</i> (Chiamata Ritornata)	<i>RETORNO</i> (Llamada de retorno)
<i>ARS</i> (Automatic Route Selection)	<i>SAA</i> (Selection de l'acheminement automatique)	<i>SAI</i> (Selez. Autom. Instradam.)	<i>SAR</i> (Seleccion automatica de rutas)
<i>forward</i> (Call Forwarding)	<i>RENVOI</i> (Renvoi)	<i>deviata</i> (Deviata)	<i>REENVIO</i> (Reenvio de llamada)
<i>cover</i> (Cover)	<i>SUPPL.</i> (Suppleance)	<i>copert.</i> (Copertura)	<i>COBER</i> (Cobertura)
<i>DND</i> (Do Not Disturb)	<i>NPD</i> (Ne pas deranger)	<i>nd</i> (Non Disturbare)	<i>NM</i> (No molestar)
<i>p</i> (Call Pickup)	<i>P</i> (Prise)	<i>a</i> (Assente)	<i>C</i> (Captura de llamada)
<i>c</i> (Cover All Calls)	<i>s</i> (Suppleance)	<i>c</i> (Copertura)	<i>c</i> (Cobertura de toda llamada)
3 of 4			

Table 81: Miscellaneous call identifier

English	French	Italian	Spanish
<i>n</i> (Night Sta. Serv., Incoming No Answer)	<i>N</i> (Service nuit, entrant pas reponse)	<i>n</i> (Serv. Notte, Esterna Non Risposta)	<i>N</i> (Servicion noct. ext. no responde)
<i>B</i> (All Calls Busy)	<i>O</i> (Tous occupes)	<i>O</i> (Tutte Occupate)	<i>O</i> (Todas ocupadas)
<i>f</i> (Call Forwarding)	<i>R</i> (Renvoi)	<i>d</i> (Deviata)	<i>R</i> (Reenvio de llamada)
<i>b</i> (Cover Busy)	<i>o</i> (Suppleance occupee)	<i>o</i> (Copertura per Occupato)	<i>o</i> (Cobertura ocupada)
<i>d</i> (Cover Don't Answer)	<i>n</i> (Suppleance pas de reponse)	<i>n</i> (Copertura per Non Risposta)	<i>n</i> (Cobertura sin respuesta)
<i>s</i> (Send All Calls)	<i>E</i> (Envoi tous appels)	<i>r</i> (Rinvio)	<i>E</i> (Envio de toda llamada)
4 of 4			

Table 82: User identifiers

English	French	Italian	Spanish	Identifier
OPERATOR	TELEPHONIST E	OPERATORE	OPERADORA	Attendant
CONFERENCE	CONFERENCE	CONFERENZ A	CONFERENCIA	Conference Call
EXT	POSTE	DER	EXTENSION	Extension
PAGING	PAGING	PAGING	PAGING	Paging (cannot be translated)
OUTSIDE CALL	APPEL EXT.	ESTERNA	LLAMADA EXT.	Trunk Group
UNKNOWN NAME	INTROUVABL E	NOME SCONOSC.	DESCONOCID O	Unknown

Table 83: Property Management System interface

English	French	Italian	Spanish
<i>CHECK IN - Ext:</i>	<i>ENREGISTREMENT - POSTE:</i>	<i>CHECK IN - Tel:</i>	<i>REGISTRARSE - EXTENSION:</i>
<i>CHECK IN: ROOM ALREADY OCCUPIED</i>	<i>ENREGISTREMENT: CHAMBRE OCCUPEE</i>	<i>CHECK IN: CAMERA OCCUPATA</i>	<i>REGISTRARSE: HABITACION OCUPADA</i>
<i>CHECK IN COMPLETE</i>	<i>ENREGISTREMENT EFFECTUE</i>	<i>CHECK IN COMPLETATO</i>	<i>REGISTRO TERMINADO</i>
<i>CHECK IN FAILED</i>	<i>ECHEC D'ENREGISTREMENT</i>	<i>CHECK IN ERRATO</i>	<i>REGISTRARSE: FALLIDO</i>
<i>CHECK OUT - Ext:</i>	<i>DEPART - POSTE:</i>	<i>CHECK OUT - Tel:</i>	<i>PAGAR LA CUENTA - EXTENSION:</i>
<i>CHECK OUT COMPLETE: MESSAGE LAMP OFF</i>	<i>DEPART: PAS DE MESSAGES</i>	<i>CHECK OUT COMPLETATO: NESSUN MESSAGGIO</i>	<i>PAGO TERMINADO: NINGUN MENSAJE</i>
<i>CHECK OUT COMPLETE: MESSAGE LAMP ON</i>	<i>DEPART: MESSAGES</i>	<i>CHECK OUT COMPLETATO: MESSAGGI IN ATTESA</i>	<i>PAGO DE CUENTA TERMINADO: MENSAJES</i>
<i>CHECK OUT FAILED</i>	<i>ECHEC PROCEDURE DE DEPART</i>	<i>CHECK OUT ERRATO</i>	<i>PAGAR LA CUENTA: FALLIDO</i>
<i>CHECK OUT: ROOM ALREADY VACANT</i>	<i>DEPART - CHAMBRE INOCCUPEE</i>	<i>CHECK OUT: CAMERA NON OCCUPATA</i>	<i>PAGAR LA CUENTA: HABITACION VACANTE</i>
<i>MESSAGE LAMP OFF</i>	<i>PAS DE MESSAGES</i>	<i>NESSUN MESSAGGIO IN ATTESA</i>	<i>LUZ DE MENSAJE APAGADA</i>
<i>MESSAGE LAMP ON</i>	<i>MESSAGES</i>	<i>MESSAGGI IN ATTESA</i>	<i>LUZ DE MENSAJE ENCENDIDA</i>
1 of 2			

Table 83: Property Management System interface

<i>English</i>	<i>French</i>	<i>Italian</i>	<i>Spanish</i>
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

2 of 2

Table 84: Security Violation Notification

<i>English</i>	<i>French</i>	<i>Italian</i>	<i>Spanish</i>
Barrier Code Violation	VIOLATION DU CODE D'ENTREE	VIOLAZIONE DI CODICI DE TAGLIO	VIOLACION CONDIGO LIMITE
Login Violation	VIOLATION DE L'ACCES A L'ADMINISTRATION	VIOLAZIONE 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	VIOLAZIONE DEL CODICE D'AUTORIZZAZIONE	VIOLACION DE CODIGO DE AUTORIZACION

Table 85: Stored number

<i>English</i>	<i>French</i>	<i>Italian</i>	<i>Spanish</i>
NO NUMBER STORED	AUCUN NUMERO EN MEMOIRE	NESSUN NUMERO IN MEMORIA	NINGUN NUMERO ALMACENADO

Table 86: Special codes

English	French	Italian	Spanish
m (Mark)	<i>M</i> (Marquer)	<i>m</i> (Marcato)	<i>M</i> (Marca)
p (Pause)	<i>P</i> (Pause)	<i>p</i> (Pausa)	<i>P</i> (Pausa)
s (Suppress)	<i>S</i> (Supprimer)	<i>s</i> (Soppresso)	<i>S</i> (Suprimir)
w (Wait)	<i>A</i> (Attendre)	<i>a</i> (Attesa)	<i>E</i> (Espera)
W (Indefinite Wait)	<i>a</i> (Attendre)	<i>A</i> (Attesa)	<i>e</i> (Espera)

Station Hunting

Table 87: Calling party display

English	French	Italian	Spanish
<i>HUNT</i>	<i>Routage</i>	<i>Ricerca</i>	<i>Busqueda</i>

Table 88: Hunt-to station display

English	French	Italian	Spanish
<i>h</i>	<i>r</i>	<i>r</i>	<i>b</i>

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 89: Time-of-Day routing messages

<i>English</i>	<i>French</i>	<i>Italian</i>	<i>Spanish</i>
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	INSTRADAMENT O 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 90: 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 91: Transfer messages

English	French	Italian	Spanish
TRANSFER COMPLETED	TRANSFERT EFFECTU	TRASFERIMENT O COMPLETATO	TRANSFERENCI A REALIZADA

Mapping enhanced display characters

Use the following tables 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

US English to Russian characters			
Russian	US English	Russian	US English
space	space	Й	Q
Ф	A	К	R
И	B	Ы	S
С	C	Е	T
В	D	Г	U
У	E	М	V
А	F	Ц	W
П	G	Ч	X
Р	H	Н	Y
Ш	I	Я	Z
О	J	Х	[
Л	K	Ъ	]
Д	L	Ж	;
Ь	M	Э	\
Т	N	Ъ	,
Щ	O	Ю	.
З	P		

US English to Japanese characters

Japanese	US English	Japanese	US English
space	space	コ	:
。	!	サ	;
「	“	シ	<
」	#	ヌ	=
.	\$	セ	>
。	%	ソ	?
ヲ	&	タ	@
ア	'	チ	A
イ	(	フ	B
ハ	)	テ	C
エ	*	ト	D
サ	+	ナ	E
ナ	,	ニ	F
エ	-	ク	G
ウ	.	ヘ	H
ッ	/	ノ	I
ー	0	ハ	J
ア	1	ヒ	K
イ	2	フ	L
ウ	3	ヘ	M
エ	4	ナ	N
オ	5	ヲ	O
カ	6	シ	P
キ	7	ソ	Q

Telephone Display

Detailed description of Telephone Display

Japanese	US English	Japanese	US English
ㇶ	8	ㇷ	R
ㇸ	9	ㇹ	S
ㇺ	[	ㇻ	T
ㇽ	\	ㇼ	U
ㇾ	]	ㇽ	V
	^	ㇿ	W
ㇿ	-	ㇾ	X
		ㇿ	Y
		ㇿ	Z

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.

European	US English	European	US English
í	space	ø	=
i	!	æ	>
ã	“	Ó	?
á	#	î	@
à	\$	â	A
ú	%	â	B
ù	&	â	C
é	‘	û	D
è	(	û	E
ç	)	è	F
†	*	é	G
Ď	+	ê	H
ý	,	Ç	I
ž	-	ř	J
ö	.	Đ	K
ò	/	ÿ	L
í	0	ñ	M
İ	1	Æ	N
Ã	2	ö	O
Á	3	î	P
À	4	Å	Q
Ú	5	Ä	R

European	US English	European	US English
Ù	6	Â	S
É	7	Û	T
È	8	Ü	U
Č	9	Ê	V
Ł	:	Ê	W
đ	;	Ë	X
Ÿ	<	Š	Y
Ř	Z	à	r
ˆ	[	Ǻ	s
Ÿ	\	û	t
Ŋ	]	ü	u
Ň	^	ě	v
ō	_	Ë	w
ì	'	Ğ	x
ä	a	Ş	y
ß	b	ř	z
û	d	þ	{
Ü	e	ž	
ě	f	ň	}
ë	g		
ğ	h		
š	i		
Ť	j		
č	k		
ž	l		
ň	m		
ñ	n		
ö	o		
í	p		
Ǻ	q		

US English to Ukrainian characters

Ukrainian	US English	Ukrainian	US English
space	space	й	Q
Ф	A	К	R
И	B	:	S
С	C	Е	T
В	D	г	U
У	E	М	V
А	F	Ц	W
П	G	ч	X
Р	H	Н	Y
Ш	I	Я	Z
О	J	Х	[
Л	K	І	]
д	L	Ж	;
б	M	Є	\
т	N	Б	,
щ	O	ю	.
з	P		

Hardware requirements for Telephone Display

The Telephone Display feature requires the following hardware:

- Telephones that can display the desired languages. See your Avaya representative for details.

Administering Telephone Display

This section describes the screens that you use to administer the Telephone Display feature.

Screens for administering Telephone Display

Screen name	Purpose	Fields
<i>Date/Time Mode and Formats - English</i>	Set up the appearance of the date and time displays.	All
<i>Date/Time Mode and Formats - French, Italian, Spanish, User-Defined, and Unicode</i>	Set up the appearance of the date and time displays.	All
<i>Language Translations</i>	Enter translations in a user-defined language.	All
<i>Leave Word Calling Format - English</i>	Set up the appearance of Leave Word Calling displays.	All
<i>Leave Word Calling Format - French, Italian, Spanish, User-Defined, and Unicode</i>	Set up the appearance of Leave Word Calling displays.	All

Reports for Telephone Display

The following reports provide information about the Telephone Display feature:

- None

Considerations for Telephone Display

This section provides information about how the Telephone Display feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Telephone Display under all conditions. The following considerations apply to Telephone Display:

- None

Interactions for Telephone Display

This section provides information about how the Telephone Display feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Telephone Display in any feature configuration.

- Distributed Communications System (DCS)
Trunk group and attendant information that is 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 that is administered on the **Trunk Group** screen.
- Single-Digit Dialing and Mixed-Station Numbering
If the system dial plan uses prefixed extensions, the prefix is not displayed when the extension is displayed. Users can use the Return Call button to dial prefixed extensions, because the system dials the prefix, even though the prefix is not displayed.

The following interactions pertain to Enhanced Telephone Display:

- Adjunct Switch Application Interface (ASAI) and related adjuncts
Information that Communication Manager sends 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, “2<@^.”
- INTUITY AUDIX Voice Power/Audix Voice Power Lodging
Not supported.
- Data Call Setup
Not supported.
- Distributed Communications Systems (DCS)
All switch nodes in a DCS network must have the same software load installed on each server or media server, must have the enhanced characters enabled, and must have telephones with the same firmware type.
- ECMA and QSIG Networking
Information must be sent between the servers that run Communication Manager.
- 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
You must use telephones that support enhanced characters. If telephones do not use enhanced characters, the software might clear the display, or display information incorrectly.

Tenant Partitioning

Use the Tenant Partitioning feature to provide telecommunications services to multiple independent groups of users through a single server that runs Avaya Communication Manager. The Tenant Partitioning feature usually provides these services from a single provider to multiple tenants of an office complex. Each tenant appears to have a dedicated Communication Manager, even though the tenants share the same Communication Manager. You can also use this feature to provide group services, such as departmental attendants, on a single-customer server that runs Communication Manager.

The Tenant Partitioning feature supports the following capabilities:

- Multiple Music-on-Hold

Detailed description of Tenant Partitioning

This section provides a detailed description of the Tenant Partitioning feature.

The Tenant Partitioning feature is not available with Offer B.

The Tenant Partitioning feature provides the following services to tenants:

- Telephone equipment
- Building wiring
- Public network and private network access
- Attendant services

You can also use the Tenant Partitioning feature to assign a separate music source to each tenant partition that plays when a user or attendant places a caller on hold.

NOTE:

If you use equipment that rebroadcasts music or other copyrighted materials, you might be required to obtain a copyright license from, or pay fees to, a third party. A Magic-on-Hold system does not require such a license. You can purchase a Magic-on-Hold system from Avaya Inc., or from an Avaya business partner.

Proper administration can protect tenant resources, including trunking facilities, and all other Communication Manager endpoints from unauthorized access by other tenants.

For the Tenant Partitioning feature to function properly in your system, you must ensure that:

- All tenants can call and be called by tenant 1. This is the system default. If you change this default, some call types fail. For example, dial 0 fails, as do security violation calls and automatic circuit assurance calls.
- 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 entity, such as an endpoint or a virtual endpoint, to which you assign a Class of Restriction (COR). You do not assign a tenant partition number to an authorization code or to fixed-assignment virtual endpoints.

You must specify an attendant group for each tenant that you define, even if the attendant group does not have an assigned console. You must also assign an attendant console to a tenant partition, and you must assign a group number to the attendant console.

Partitioning tenants

The system has a default of one universal tenant for the system. This tenant is in partition 1, and is usually the service provider. Another system default is that the tenant in partition 1 has access to all facilities in the system, and that all other tenants can access the tenant in partition 1. The tenant in partition one is usually referred to as tenant 1.

The service provider creates additional partitions that are based on tenant requirements. When you create tenant partitions, remember that:

- You can assign each Communication Manager endpoint to only one tenant partition. And, you must pass each endpoint to a partition. For example, you must assign each telephone, attendant console, trunk, and virtual endpoint. Virtual endpoints include listed directory numbers and virtual directory number (VDN) to a tenant partition.
- Most tenant partitions are separate units. By default, the system does not allow tenants, except tenant 1, to access telephones or trunk facilities that belong to other tenants. However, you can change this system default. You can give explicit permission for a tenant to access another tenant. For example, you can allow tenant 6 to call only tenant 9 and tenant 16.

NOTE:

If a tenant has permission to call another tenant, the tenant has access to all the endpoints that belong to the other tenant. For example, if tenant 6 has permission to call tenant 9, tenant 6 can also use any trunk facilities in tenant partition 9.

- The system can use tenant partitioning restrictions to block calls between two users in the system. However, either user can use the Direct Inward Dialing (DID) extension of the other user to call over the public network.
- If tenants want to share some, but not all, facilities, you must group the shared facilities into a separate partition. For example, if two tenants share a trunk, but do not have direct system access to call each other, put the trunk in its own partition. Both tenants can then access the trunk.

You must also consider the constraints and the requirements of access control, attendant services, music sources on hold, and network route selection when you establish or assign partitions.

Access control

Tenant-to-tenant access restrictions limit some features, such as Call Coverage. For example, the coverage of tenant 2 includes a telephone from tenant 3 in its coverage path. If tenant 4 has permission to call tenant 2, but does not have permission to call tenant 3, a call from tenant 4 to tenant 2 skips the tenant 3 coverage point.

You might also want to set up tenants with special access privileges. For example, you might give a restaurant in an office complex permission to receive calls from any other tenant. You might also want tenants to have permission to call, or be called by, building security, or those who administer and troubleshoot Communication Manager.

You can also assign all central office (CO) trunks to one tenant partition. All other tenants can then access the CO trunks.

Attendant services

With Tenant Partitioning, you can administer personalized attendant services for each tenant.

The system provides one principal attendant, and either one night attendant or one day and night attendant, for each 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, when someone presses a queue-status button, the system displays the status of the attendant-group queue. The total number of calls queued for all tenants cannot exceed the system limit.

Attendant groups can serve more than one tenant, if the two tenants have permission to access each others facilities. For example, can use the facilities of one tenant to extend a call to another tenant, if the tenant has permission to access the facilities of the other tenant.

Each tenant can have a designated night-service station. When a night attendant is unavailable, the system directs calls to an attendant group that is in night service, to the night-service station of the appropriate tenant. When someone places an attendant group into night service, all trunk groups and hunt groups that belong to tenants that the attendant group serves, 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 exclusive access to the following entities:

- Listed directory number (LDN) night destination
- Trunk answer on any station (TAAS) port
- 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 group. The system directs any valid user attempt to access the trunk group to the attendant group that serves the tenant.

Network route selection

You can place trunk groups that belong to different tenants in the same route pattern. Calls that the system routes to that pattern select the first trunk group in the pattern that has caller access permission.

Tenant Partitioning examples

The following example describes 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 through 15 to individual businesses in the complex. You maintain the system-default restrictions for these tenants. These restrictions specify that tenants cannot access the telephones, the trunk facilities, or the other Communication Manager endpoints that belong to other tenants.

You assign tenant partition 16 to the restaurant in the building complex. You give all tenants permission to call the restaurant. However, to prevent access to trunks and other facilities that belong to other tenants, you do not allow the restaurant to call the other tenants.

You assign tenant partition 17 to all the CO trunk groups. You give all tenants permission to call tenant 17.

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, and you deny all other tenants access to partition 18. To prevent toll fraud, you do not allow tenant 18 to place calls to tenant 18.

All tenants can use the same Automatic Route Selection (ARS) route pattern. In this example, the trunk for tenant partition 18, the private trunk that tenants 3 and 7 share, is first in the route pattern. Tenant partition 17 is second in the route pattern. The system routes calls that tenants 3 and 7 make to partition 18 and then, as a second option, to partition 17. The system routes all other tenants directly to partition 17, because you deny all other tenants access to partition 18.

Assign the facilities that the tenants do not share to the tenant partition that the facilities serve. These facilities can include trunk groups, VDNs, telephones, attendant consoles, and other endpoints.

[Table 92, Calling permissions for the partitions](#), on page 1062 summarizes the calling permissions for the different tenant 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 92: Calling permissions for the partitions

Calling tenant partition number	Called tenant partition number					
	1	2, 4 through 6, and 8 through 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 cannot call the other partitions.	No	Yes	Yes	No
3, 7	Yes	No	Each partition can call itself, but cannot call the other partitions.	Yes	Yes	Yes
16	Yes	No	No	Yes	Yes	No
17	Yes	Yes	Yes	Yes	Yes	No
18	Yes	No	Yes	No	No	No

Multiple Music-on-Hold

Tenant Partitioning allows you to assign a separate music source to each tenant partition. A caller hears the music when a user places a call on hold. The tenant number that you assign to the called extension usually determines the music source that the user hears. This capability allows you to customize the music, or the messages, for the business needs of each tenant partition.

If the COR of the user extension that places the call on hold allows music-on-hold, a caller on hold hears the music source that is assigned to the partition at which the call initially terminates. For example, if the system first routes a call to an INTUITY automated attendant, and then routes the call to the appropriate tenant partition, the caller who is on hold hears the music source of the INTUITY automated attendant. The caller who is on hold does not hear the music source of the tenant partition to which the system routes the call. Likewise, if a caller in tenant partition 2 makes an out-going call that uses the trunk groups of tenant 3, the caller hears the music source that is assigned to tenant 3. If the COR of the called extension does not allow music on hold, the caller hears nothing.

The maximum number of allowed music sources is the same as the maximum number of allowed tenant partitions. More than one tenant partition can use each music source.

[Table 93, Types of music-on-hold](#), on page 1063 shows which music-on-hold types that you can assign to each tenant partition.

Table 93: Types of music-on-hold

Type	System response for a call who is on hold
none	Silence
tone	System-wide administered tone
music	This is the type of music that is associated with the administered port. The number of allowed music sources equals the number of allowed tenant partitions. Each partition can have its own music source.

Hardware requirements for Tenant Partitioning

The Tenant Partitioning feature requires the following hardware:

- None

Administering Tenant Partitioning

The following steps are part of the administration process for the Tenant Partitioning feature:

- [Defining a tenant partition](#)
- [Assigning a tenant partition number to an access telephone](#)
- [Assigning a tenant partition number to an agent login ID](#)
- [Assigning a tenant partition number to an announcement](#)
- [Assigning a tenant partition number to an attendant](#)
- [Assigning a tenant partition number to a data module](#)
- [Assigning a tenant partition number to a hunt group](#)
- [Assigning a tenant partition number to a loudspeaker paging zone](#)
- [Assigning a tenant partition number to the remote access extension](#)

- [Assigning a tenant partition number to a user extension](#)
- [Assigning a tenant partition number to a terminating extension group](#)
- [Assigning a tenant partition number to a trunk group](#)
- [Assigning a tenant partition number to a vector directory number](#)
- [Assigning the sources of music for the tenant partitions](#)

This section describes:

- Any prerequisites for administering the Tenant Partitioning feature
- The screens that you use to administer the Tenant Partitioning feature
- Complete administration procedures for the Tenant Partitioning feature

Prerequisites for administering Tenant Partitioning

You must complete the following actions before you can administer the Tenant Partitioning feature:

- Ensure that the **Tenant Partitioning?** field on the **Optional Features** screen is set to **y**. If the **Tenant Partitioning?** field is set to **n**, your system is not enabled for the Tenant Partitioning feature. Contact your Avaya representative for assistance before you continue with this procedure.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

Screens for administering Tenant Partitioning

Screen name	Purpose	Fields
Access Endpoint	Assign a tenant partition number to an access endpoint.	TN
Agent LoginID	Assign a tenant partition number to an agent login ID.	TN
Announcements/Audio Sources	Assign a tenant partition number to an announcement that is assigned to an extension.	TN
Attendant Console	Assign the group number and the tenant partition number to an attendant.	• Group • TN
Data Module	Assign a tenant partition number to a data module.	TN
Feature-Related System Parameters	Enable the Tenant Partitioning feature for your system.	Tenant Partitioning?
Hunt Group	Assign a tenant partition number to a hunt group.	TN

Screen name	Purpose	Fields
<i>Loudspeaker Paging</i>	Assign a tenant partition number to each loudspeaker paging zone.	TN
<i>Music Sources</i>	Assign the sources of the music for the system.	All
<i>Remote Access</i>	Assign a tenant partition number to the remote access extension.	TN
<i>Station</i>	Assign a tenant partition number to a user extension.	TN
<i>Tenant</i>	Define a tenant to the system.	All
<i>Terminating Extension Group</i>	Assign a tenant partition number to a Terminating Extension Group (TEG).	TN
<i>Trunk Groups</i>	Assign a tenant partition number to a trunk group.	TN
<i>Vector Directory Number</i>	Assign a tenant partition number to a vector directory number (VDN).	TN

Defining a tenant partition

To assign a music source to a tenant partition:

- 1 Type **change tenant *n***, where *n* is the tenant that you want to change. Press **Enter**.

The system displays the **Tenant** screens ([Figure 277, Tenant screen](#), on page 1065), ([Figure 278, Tenant screen](#), on page 1066).

Figure 277: Tenant screen

```

change tenant
                                Tenant 18

      Tenant Description: _____
      Attendant Group: 1
Ext Alert Port (TAAS): _____ Ext Alert (TAAS) Extension: ____*
      Night Destination: _____
      Music Source: 1
  
```

Figure 278: Tenant screen

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

The **Tenant** field is a display-only field. This field contains the tenant number that you typed on the command line.

- 2 In the **Tenant Description** field, type a description of the tenant. You can type 40 characters. You can leave the field blank, but a description helps you identify the tenant when you change other information about the tenant.

- 3 In the **Attendant Group** field, type the number of the attendant group for the tenant partition. The system assigns the number 1 as the default for the **Attendant Group** field.

The default for the system is that all attendant groups exist. However, the attendant group is empty if you do not assign consoles to the attendant group.

- 4 In the **Ext Alert Port (TAAS)** field, perform one of the following actions:

- If trunk answer from any station (TASS) alert port information does not exist, type an **x**.
- If TASS alert port information exists, enter the 7-character port number.

The circuit pack must be installed and defined to the system before you can refer to the circuit pack in the **Ext Alert Port (TAAS)** field. The port type and the object type must be consistent. You can assign the port to only one tenant. Use the information in [Table 94, Port information for the Ext Alert Port \(TAAS\) field](#), on page 1067 to construct the 7 character port number.

Table 94: Port information for the Ext Alert Port (TAAS) field

Characters	Meaning	Value
1-2	Cabinet Number	01 through 03
3	Carrier	A through E
4-5	Slot Number	00 through 20
6-7	Circuit Number	01 through 31

5 In the **Night Destination** field, type the night service station extension, if you want night service for the tenant. Type an extension that already exists in the system.

6 In the **Ext Alert (TAAS) Extension** field, type an extension that already exists in the system.

Note that, the system displays the **Ext Alert (TAAS) Extension** field if you type an **x** in the **Ext Alert Port (TAAS)** field.

7 In the **MUSIC** field, type the source of the music or the tone for the tenant partition.

8 Press **Enter** to save your changes.

9 Page through the Tenant screens until you see the **Calling Permission** area.

Note that the **Tenant** field is a display only field. This field contains the tenant number that you entered on the command line.

10 In the numbered fields, perform one of the following actions:

- Type **y** to enable calling permission between the tenant that you entered in the command line and any other tenant in the system.
- Type **n** to disable calling permission between the tenant that you entered in the command line and any other tenant in the system.

The system default for the calling permissions are to:

- Allow the tenant to call itself
- Allow the tenant to call tenant 1
- Turn off all other calling permissions between tenants

Assigning a tenant partition number to an access telephone

To assign a tenant partition number to an access telephone:

1 Type **change access-endpoint *n***, where ***n*** is the extension of the access telephone. Press **Enter**.

The system displays the **Access Endpoint** screen ([Figure 279, Access Endpoint screen](#), on page 1068).

Figure 279: Access Endpoint screen

change access-endpoint 30001	ACCESS ENDPOINT	Page 1 of 1
Extension: 30001	(Starting) Port: _____	
Communication Type: voice-grade-data	Name: _____	
COR: 1	COS: 1	
TN: 1	ITC: restricted	

- 2 In the TN field, type the tenant partition number for the access endpoint. The valid entries for this field are **1** to **100**.
- 3 Press **Enter** to save your changes.

Assigning a tenant partition number to an agent login ID

To assign a tenant partition number to an announcement:

- 1 Type **change agent loginid *n***, where *n* is the number of the agent login ID that you want to change. Press **Enter**.

The system displays the *Agent LoginID* screen ([Figure 280, Agent LoginID screen](#), on page 1068).

Figure 280: Agent LoginID screen

change agent-loginID 100	AGENT LOGINID	Page 1 of 2
Login ID: 100	AAS? n	
Name:	AUDIX? n	
TN: 1	LWC Reception: spe	
COR: 1	LWC Log External Calls? n	
Coverage Path:	AUDIX Name for Messaging:	
Security Code:	LoginID for ISDN Display? n	
	Password:	
	Password (enter again):	
	Auto Answer: station	
WARNING: Agent must log in again before skill changes take effect		

- 2 In the TN field, type the tenant partition number for the agent login ID. The system assigns the number 1 as the default for a tenant partition number.
- 3 Press **Enter** to save your changes.

Assigning a tenant partition number to an announcement

To assign a tenant partition number to an announcement:

- 1 Type **change announcements**. Press **Enter**.

The system displays the *Announcements/Audio Sources* screen ([Figure 281, Announcements/Audio Sources screen](#), on page 1069).

Figure 281: Announcements/Audio Sources screen

ANNOUNCEMENTS/AUDIO SOURCES										Page 1 of X	
Ext.	Type	COR	TN	Name	Q	QLen	Pro	Rate	Port		
1: 1701_	analog_	1_	1_	busy	y	1000	N/A	N/A	1B0404		
2: 1702_	analog_	1_	1_	out of service		N/A	N/A	N/A	1B1701		
3: 1703_	integrated	1_	1_	please dial	n	N/A	n	32	1B18		
4: 1704_	aux-trunk_	1_	1_	call-back	y	20	N/A	N/A	2C0902		
5: _____	_____	1_	1_	_____	n						
6: _____	_____	1_	1_	_____	n						
7: _____	_____	1_	1_	_____	n						
8: _____	_____	1_	1_	_____	n						
9: _____	_____	1_	1_	_____	n						
10: _____	_____	1_	1_	_____	n						
11: _____	_____	1_	1_	_____	n						
12: _____	_____	1_	1_	_____	n						
13: _____	_____	1_	1_	_____	n						
14: _____	_____	1_	1_	_____	n						
15: _____	_____	1_	1_	_____	n						
16: _____	_____	1_	1_	_____	n						

- 2 In the TN field, type the tenant partition number for the extension. The system assigns the number 1 as the default for a tenant partition number.
- 3 Press **Enter** to save your changes.

Assigning a tenant partition number to an attendant

To assign the tenant partition number to an attendant:

- 1 Type **change attendant *n***, where *n* is the number of the attendant that you want to change.

The system displays the Attendant Console screen, [Figure 282, Attendant Console screen](#), on page 1070.

Figure 282: Attendant Console screen

```

change attendant 1                                     Page 1 of 4
               ATTENDANT CONSOLE 1

      Type: console      Name: Gertrude Attendant 64B0824
      Extension: 70199    Group: 1      Auto Answer: none
      Console Type: principal  TN: 1      Data Module? n
      Port: 64B0824      COR: 1      Disp Client Redir? n
                        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:                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 **Group** field, type the group number for the Attendant. The valid entries for this field are **1** to **28**.
- 3** In the **TN** field, type the tenant partition number for the Attendant. The valid entries for this field are **1** to **100**.
- 4** Press **Enter** to save your changes.

Assigning a tenant partition number to a data module

To assign a tenant partition number to a data module:

- 1** Type **change data-module *n***, where *n* is the extension of the data module. Press **Enter**.
The system displays the **Data Module** screen ([Figure 283, Data Module screen](#), on page 1071).

Figure 283: Data Module screen

change data-module 30

DATA MODULE

Page 1 of 2

Data Extension: 30

Type: data-line____

Port: _____

ITC: restricted__

Name: 27

COS: 1

COR: 1

TN: 1

BCC:

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

- 2 In the TN field, type the tenant partition number for the data module. The valid entries for this field are **1** to **100**.
- 3 Press **Enter** to save your changes.

Assigning a tenant partition number to a hunt group

To assign a tenant partition number to a hunt group:

- 1 Type **change hunt-group *n***, where *n* is the hunt group number to which you want to assign a partition number. Press **Enter**.

The system displays the **Hunt Group** screen ([Figure 284, Hunt Group screen](#), on page 1072).

Figure 284: Hunt Group screen

```

change hunt-group 4                                     Page 1 of X
                                     HUNT GROUP

      Group Number: 4__
      Group Name: _____
      Group Extension: _____
      Group Type: _____
      TN: _____
      COR: _____
      Security Code: _____
      ISDN Caller Disp: _____

      ACD? _
      Queue? y
      Vector? _
      Coverage Path: _____
      Night Service Destination: _____
      MM Early Answer? _

      Queue Length: _____
      Calls Warning Threshold: _____ Port: x Extension: _____
      Time Warning Threshold: _____ Port: x Extension: _____
  
```

- 2 In the TN field, type the tenant partition number for the hunt group.
- 3 Press **Enter** to save your changes.

Assigning a tenant partition number to a loudspeaker paging zone

To assign a tenant partition number to a loudspeaker paging zone:

- 1 Type **change paging loudspeaker**. Press **Enter**.

The system displays the **Loudspeaker Paging** screen ([Figure 285, Loudspeaker Paging screen](#), on page 1072).

Figure 285: Loudspeaker Paging screen

```

change paging loudspeaker                               Page 1 of 1
                                     LOUDSPEAKER PAGING

      CDR? y
      Voice Paging Timeout (sec):
      Code Calling Playing Cycles:

PAGING PORT ASSIGNMENTS
Zone  Port      Voice Paging      Code Calling      Location:
      TAC  COR  TN      TAC  COR  TN
1:          1          1          1      27 character LOUDSPK PAGING
2:          1          1          1      PAGING
3:          1          1          1      PAGING
4:          1          1          1      PAGING
5:          1          1          1      PAGING
6:          1          1          1      PAGING
7:          1          1          1      PAGING
8:          1          1          1      PAGING
9:          1          1          1      PAGING
ALL:          1          1          1
  
```


- 2 In the TN field, type the tenant partition number for the loudspeaker paging zone.
- 3 Press **Enter** to save your changes.

Assigning a tenant partition number to the remote access extension

To assign a tenant partition number to the remote access extension:

- 1 Type **change remote-access**. Press **Enter**.

The system displays the **Remote Access** screen ([Figure 286, Remote Access screen](#), on page 1073).

Figure 286: Remote Access screen

```

change remote-access

                                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)

```

- 2 In the TN field, type the tenant partition number for the remote access extension. The system assigns the number 1 as the default tenant partition number.
- 3 Press **Enter** to save your changes.

Assigning a tenant partition number to a user extension

To assign a tenant partition number to a user extension:

- 1 Type **change station *n***, where *n* is the extension to which you want to assign a tenant partition number. Press **Enter**.

The system displays the **Station** screen ([Figure 287, Station screen](#), on page 1074).

Figure 287: Station screen

change station 30019		STATION	
Extension: 30019	Lock Messages? n	BCC: 0	
Type: 4612	Security Code: *	TN: 1	
Port: S04007	Coverage Path 1:	COR: 1	
Name: station 30019	Coverage Path 2:	COS: 1	
	Hunt-to Station:		
STATION OPTIONS			
Loss Group: 19	Personalized Ringing Pattern: 1		
	Message Lamp Ext: 30019		
Speakerphone: 2-way	Mute Button Enabled? y		
Display Language: english			
	Media Complex Ext:		
	IP SoftPhone? n		

- 2 In the TN field, type the tenant partition number for the user extension.
- 3 Press **Enter** to save your changes.

Assigning a tenant partition number to a terminating extension group

To assign a tenant partition number to a Terminating Extension Group (TEG):

- 1 Type **change term-ext-group *n***, where *n* is the terminating extension group number to which you want to assign a tenant partition number. Press **Enter**.

The system displays the *Terminating Extension Group* screen ([Figure 288, Terminating Extension Group screen](#), on page 1074).

Figure 288: Terminating Extension Group screen

change term-ext-group		Page 1 of 1	
TERMINATING EXTENSION GROUP			
Group Number: 1	Group Extension:		
Group Name: 27 character	Coverage Path:		
Security Code:	COR: 1		
	TN: 1		
ISDN Caller Disp:	LWC Reception: none		
AUDIX Name:	Messaging Server Name:		
GROUP MEMBER ASSIGNMENTS			
Ext	Name	Ext	Name
1: 51001	27 character name sta 51001	3:	
2:		4: 51002	27 character name sta 51002

- 2 In the TN field, type the tenant partition number for the TEG. The system assigns the number 1 as the default for a tenant partition number.
- 3 Press **Enter** to save your changes.

Assigning a tenant partition number to a trunk group

To assign a tenant partition number to a trunk group:

- 1 Type **change trunk-group *n***, where *n* is the trunk group number to which you want to assign a tenant partition number. Press **Enter**.

The system displays the **Trunk Group** screen ([Figure 289, Trunk Group screen](#), on page 1075).

Figure 289: Trunk Group screen

change trunk-group 3		Page 1 of 22
TRUNK GROUP		
Group Number: 3	Group Type: isdn	CDR Reports: y
Group Name: GERT tg3 tf Lulu tg26 PRI	COR: 2	TN: 1 TAC: 1003
Direction: two-way	Outgoing Display? y	Carrier Medium: PRI/BRI
Dial Access? y	Busy Threshold: 255	Night Service:
Queue Length: 0	Auth Code? n	TestCall ITC: rest
Service Type: tie	Far End Test Line No:	
TestCall BCC: 4		
TRUNK PARAMETERS		
Codeset to Send Display: 6	Codeset to Send National IEs: 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? n
		Digital Loss Group: 13
Incoming Calling Number - Delete:	Insert:	Format:
Bit Rate: 1200	Synchronization: async	Duplex: full
Disconnect Supervision - In? y	Out? n	
Answer Supervision Timeout: 0		

- 2 In the TN field, type the tenant partition number for the trunk group.
- 3 Press **Enter** to save your changes.

Assigning a tenant partition number to a vector directory number

To assign a tenant partition number to a vector directory number (VDN):

- 1 Type **change vdn *n***, where *n* is the VDN to which you want to assign a tenant partition number. Press **Enter**.

The system displays the **Vector Directory Number** screen ([Figure 290, Vector Directory Number screen](#), on page 1076).

Figure 290: Vector Directory Number screen

change vdn 5000	VECTOR DIRECTORY NUMBER	Page 1 of 1
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:		

- 2 In the TN field, type the tenant partition number for the VDN.
- 3 Press **Enter** to save your changes.

Assigning the sources of music for the tenant partitions

To assign the sources of music for the tenant partitions:

- 1 Type **change music-sources**. Press **Enter**.

The system displays the **Music Sources** screen ([Figure 291, Music Sources screen](#), on page 1076).

Figure 291: Music Sources screen

change music-sources				Page 1 of X
Source	Type	Music Sources 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			

The **Source** field is a display-only field.

Note that if the **Tenant Partitioning?** field on the *Optional Features* screen is set to **y**:

- The **Music/Tone on Hold** field on the *Feature-Related System Parameters* screen disappears.
- The value in the **Music/Tone on Hold** field on the *Feature-Related System Parameters* screen appears in the first **Source** field on the *Music Sources* screen.
- If the value in the **Music/Tone on Hold** field on the *Feature-Related System Parameters* screen contained **music**, the port address also appears on the *Music Sources* screen.

Note that when the **Tenant Partitioning?** field on the *Optional Features* screen is changed from **y** to **n**:

- The **Music/Tone on Hold** field reappears on the *Feature-Related System Parameters* screen.
- The system displays the value from the *Music Sources* screen, in the **Music/Tone on Hold** field on the *Feature-Related System Parameters* screen.

- 2 In the **Type** field, perform one of the following actions:
 - If you want the users to hear music, type **music**.
 - If you want the users to hear the tone-on-hold tone, type **tone**.
You can specify **tone** for only one music source.
 - If you want the users to hear nothing, type **none**.
- 3 In the **Port** field, type the auxiliary trunk address or the analog port address of the music source. You cannot enter duplicate addresses in the **Port** field.

The system displays the **Port** field only if you typed **music** in the **Type** field.
- 4 In the **Description** field, type a maximum of 20 characters that describe the source of the music.

The system displays the **Description** field, only if you typed **music** or **tone** in the **Type** field.
- 5 Press **Enter** to save your changes.

Reports for Tenant Partitioning

The following reports provide information about the Tenant Partitioning feature:

- None

Considerations for Tenant Partitioning

This section provides information about how the Tenant Partitioning feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Tenant Partitioning under all conditions. The following considerations apply to Tenant Partitioning:

- None

Interactions for Tenant Partitioning

This section provides information about how the Tenant Partitioning feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Tenant Partitioning in any feature configuration.

Tenant-partition identification is not passed between servers. A network of servers that run Communication Manager does not enforce Tenant Partitioning restrictions without special administration. For example, Tenant Partitioning on a network of servers that run Communication Manager does not enforce tenant-specific tie trunks.

Administration of the following features requires special care to avoid unintended access between tenants.

- 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 central office (CO) line
- Private Networking - automatic alternate routing (AAR)
- Service Observing
- Uniform Dial Plan

Specific information about how the Tenant Partitioning feature interacts with other features follows.

- Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS)

The Tenant Partitioning feature is not the same as the Time-of-Day Plan Numbers or the Partition Groups in AAR and ARS.

However, if you use the Tenant Partitioning feature, you can use the Time-of-Day Plan Numbers and the Partition Groups in AAR and ARS.

- Attendant and Attendant Group features

The Tenant Partitioning feature creates multiple attendant groups. Attendant operations such as direct-station or direct trunk group select (DTGS) are subject to tenant-to-tenant restrictions, both at selection time and at split time.

All the calls that an attendant within an attendant group places on hold hear the music source from the attendant group.

- Attendant Control of Trunk Group Access

An attendant group controls access to the trunk groups of the tenants that the attendant group serves. An attendant does not control access to any other trunk groups.

- AUDIX and embedded AUDIX

The system applies the same tenant-to-tenant restrictions to AUDIX voice and data ports that the system applies to any other endpoints.

AUDIX can restrict one group of subscribers from sending voice mail to another tenant partition group.

Those who control the tenant partitions can:

- Create 10 different communities within each AUDIX
- Allow voice message access across the community boundaries
- Deny voice message access across the community boundaries

- Authorization Codes

Authorization codes are associated with class of restriction (COR). If you want to assign a unique set of authorization codes to a tenant, you create a unique set of CORs. You only assign the CORs to objects that are within the partition that the tenant uses.

- Automatic Wakeup

Those who control a tenant partition assign a music source to the partition. The system uses this music source as the wakeup music for users in the partition.

- Bridged Call Appearance

Assign all stations with bridged call appearances to the same tenant.

- Call Coverage

The tenant-to-tenant access restrictions apply to coverage paths. The system does not allow a call to cover to a tenant if the tenant-to-tenant access restrictions do not allow the user to call the tenant.

When you specify an attendant in a coverage path, the system accesses the attendant group of the called tenant. The system does not access the attendant group of the calling tenant.

When the system uses the Call Coverage feature for a call, and a user answers the call and then places the call on hold, the system plays the music-on-hold music that is assigned to the original called party.

- Call Detail Recording (CDR)

CDR does not report the tenant partition number of the extension or the trunk group that the system uses. You must infer the tenant partition number from the extension or the trunk-group number.

- Call Pickup

Assign all stations in a call-pickup group to 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 and Vector Directory Number (VDN)

When the system routes a call to a new destination as a result of a vector step, the caller hears the music that is assigned to the last active VDN.

While a call is in vector processing, the system uses the tenant number that is assigned to the active VDN, as determined by VDN Override, to select the music source for callers on hold. Note the following exception. If you use a *wait-time <time> hearing <extension> then <treatment 2>* command where the *<extension>* is a music source, that music source plays instead of the music source that is associated with the active VDN.

The COR that is assigned to the VDN must allow music-on-hold.

- **Call Management System (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, the tenant accesses the attendant group to which the tenant is assigned.
- **Emergency Access to the Attendant**
When a tenant dials the emergency access, the tenant accesses the attendant group to which the tenant is assigned.
- **Expert Agent Selection (EAS)**
The COR that you assign to the logical agent ID in the EAS system determines whether callers that are on hold hear music. The COR that you assign to the physical extension does not control whether callers that are on hold hear music.
- **Hunt groups**
The tenant number that you assign to the hunt group extension determines the music source that the callers to the hunt group hear while the callers are in queue or on hold.
- **Intercept Treatment**
When access to the attendant is designated as intercept treatment, the caller accesses the attendant group to which the caller is assigned.
- **Malicious Call Trace (MCT)**
The system assigns MCT extensions to tenant partition 1 as the default for the system. If MCT is enabled, any user who has permission to call tenant partition 1 can use MCT.
- **Multiple Listed Directory Numbers (LDNs)**
Assign a tenant partition to each LDN.
- **Multiple Audio and Music Sources for Vector Delay**
When you specify the audio source by a ***wait-time <time> hearing <treatment>*** vector step, the audio source that is assigned to the tenant number of the active VDN is the audio source that plays.

If you use a ***wait-time <time> hearing <extension> then <treatment 2>*** command where the ***<extension>*** is an audio source, that audio source plays instead of the audio source that is associated with the active VDN. For information on administering multiple audio sources, see the *Avaya Communication Manager Contact Center Call Vectoring and Expert Agent Selection (EAS) Guide*.
- **Music-on-Hold Access**
You can assign a unique source for music to each tenant.
- **Night Service**
Each tenant can have a listed directory number (LDN) night destination, a trunk answer on any station (TAAS) port, or a night attendant that only the tenant uses.
- **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 the trunk group. You must infer the tenant partition number from the extension or trunk-group number.
- **Uniform Dial Plan (UDP)**
If a UDP is in place between servers, the system does not pass tenant partition identification between the servers. The system does not enforce tenant-partition restrictions between the servers unless you provide special administration.

Tenant Partitioning restrictions do not override COR restrictions. COR restrictions are independent of tenant partitions.

Temporary Bridged Appearance

Use the Temporary Bridged Appearance feature to allow multiappearance telephone users in a Terminating Extension Group (TEG) or Personal Central Office Line (PCOL) group to bridge onto an existing group call. If the Call Pickup feature is used to answer the call, 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 of Temporary Bridged Appearance

This section provides a detailed description of the Temporary Bridged Appearance feature.

An incoming call to a Terminating Extension Group (TEG) or Private Central Office Line (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 does not answer the call originally, this individual can bridge onto the call. The answering party does not have to transfer the call.

A call to an individual can be answered by a member of a call pickup group. While the call is still connected, the called party can bridge onto the call, and the answering party hangs up.

Call Coverage provides redirection of calls to alternate answering positions or covering users. A temporary bridged appearance is maintained at the called telephone.

The called party can answer the call at any time, even if the call 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 that is maintained on the call. When the consult call is finished, the temporary bridged appearance is removed.

Stations that usually have a temporary bridged appearance with their coverage point do not have a temporary bridged appearance if the coverage point is INTUITY AUDIX.

Hardware requirements for Temporary Bridged Appearance

The Temporary Bridged Appearance feature requires the following hardware:

- None

Administering Temporary Bridged Appearance

This section describes the screens that you use to administer the Temporary Bridged Appearance feature.

Screens for administering Temporary Bridged Appearance

Screen name	Purpose	Fields
<i>Feature-Related System Parameters</i>	Enable the Temporary Bridged Appearance feature.	Temporary Bridged Appearance on Call Pickup

Reports for Temporary Bridged Appearance

The following reports provide information about the Temporary Bridged Appearance feature:

- None

Considerations for Temporary Bridged Appearance

This section provides information about how the Temporary Bridged Appearance feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Temporary Bridged Appearance under all conditions. The following considerations apply to Temporary Bridged Appearance:

- Temporary Bridged Appearance allows a party to bridge onto a call. The answering party does not have to transfer the call, which is convenient and saves time.
- Temporary Bridged Appearance does not provide the capability to originate calls, or the capability to answer another party's calls. The Bridged Call Appearance feature provides these capabilities.
- 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 for Temporary Bridged Appearance

This section provides information about how the Temporary Bridged Appearance feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Temporary Bridged Appearance in any feature configuration.

- Call Coverage

Calls that are 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, all three parties hear the bridging tone.

- Consult

Consult calls use the temporary bridged appearance that is 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 had a temporary bridged appearance and the call is then conferenced or transferred, and is redirected to coverage again, a temporary bridged appearance is not maintained at the conferenced-to or the 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.

Terminal Translation Initialization

Use the Terminal Translation Initialization (TTI) feature to merge an X-port extension to a valid port, or to separate an extension from a port.

Detailed description of Terminal Translation Initialization

This section provides a detailed description of the Terminal Translation Initialization (TTI) feature.

Use TTI to:

- Merge an X-port extension to a valid port. To merge an X-port extension to a valid port, enter the system-wide TTI security code, and then the extension from a telephone that is connected to the valid port.
- Separate an extension from the port to which the extension is assigned. To separate the extension from the port, enter the required numbers. When you enter the required numbers, the extension is 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.

You usually use TTI to move telephone. However, you can also use TTI to connect and move attendants and data modules.

Attendant consoles

You must assign an extension to the attendant console, if you want the attendant to use TTI.

TTI port translations are the same for both digital telephones and attendant consoles. To merge a digital TTI voice port and an attendant console, you must:

- 1 Administer the attendant console as an X port.
- 2 Plug a digital telephone into the jack that is assigned to the attendant console.
- 3 Enter the TTI merge digit sequence at the digital telephone.
- 4 Unplug the digital telephone.
- 5 Plug the attendant console into the jack.

You can separate an attendant console from its port only through administration. A TTI separate request from an attendant console gives the user intercept treatment.

Data modules

The system provides status information during the TTI merge, and the separate operations on a telephone that is connected to a data module. If the **TTI State** field on the *Feature-Related System Parameters* screen is set to **data**, the system displays the status information. If the **TTI State** field on the *Feature-Related System Parameters* field is set to **voice**, the system generates tones to indicate status.

For a stand alone data module, you enter the TTI merge and separate digit sequence on one line at a DIAL prompt:

- DIAL: <*TTI feature access code*><*TTI security code*><*extension*>

The system does not generate separate prompts for the TTI security code and the extension.

Voice and data telephones

The system process a telephone with a data terminal (DTDM) as a telephone in the TTI merge and separation sequence operations. The DTDM is merged, with and separated from, its hardware translation at the same time that the telephone is merged or separated. You can only start the TTI merge and separate sequence only through the telephone for DTDMs. You cannot start the sequences through the data port.

ISDN-BRI telephones

You use the same TTI separation sequence for Automatic-TEI SPID-initializing BRI telephones that you use for other telephones. However, the merge sequence is different.

- Separation sequence
 - 1 Feature access code (FAC)
 - 2 Security code
 - 3 Extension
- Merge sequence
 - 1 Connect the telephone to any port to get power.
 - 2 Program the service profile identifier (SPID) to the extension with which the telephone is to be merged.
 - 3 Unplug the telephone.

You must unplug the telephone, even if the telephone is connected to the intended port.
 - 4 Connect the telephone to the intended port.

The intended port must indicate Equipment Type: TTI Port.
 - 5 Listen for the dial tone.
 - If you hear dial tone, the merge is complete.
 - If you do not hear dial tone, the SPID of the telephone is an unavailable extension.

You can dial the TTI merge sequence for BRI telephones only if a user separates a BRI extension from its telephone, and then wants to reassociate the telephone to the same extension. Note that you cannot use the system access terminal (SAT) to put an **x** in the **Port** field of a BRI *Station* record that is still connected to the server that runs Avaya Communication Manager. You must use the TTI separation sequence from the telephone.

Analog queue warning ports and external alert ports

You can administer the analog queue warning port that is used for hunt groups, and the external alert port, with an **x** in the **Port** field. You can use TTI to merge these extensions to an analog port. You must perform the merge at an analog set, and then unplug the analog set from the port. You cannot use TTI to separate these extensions from their port location. A TTI separate request from one of these ports gives you intercept treatment.

Security

SECURITY ALERT:

If you do not manage TTI carefully, unauthorized use of this feature can cause you security problems. For example, someone who knows the TTI security code can disrupt normal business functions by separating telephones or data terminals. You can help protect against unauthorized use of TTI by frequently changing the TTI security code. You can further enhance system security by removing the feature access code (FAC) from the system when the FAC is not needed. Consult the *Avaya Products Security Handbook* for additional information to secure your system, and to find out about regularly obtaining updated security information.

Hardware requirements for Terminal Translation Initialization

The Terminal Translation Initialization feature requires the following hardware:

- None

Administering Terminal Translation Initialization

This section describes the screens that you use to administer the Terminal Translation Initialization feature.

Screens for administering Terminal Translation Initialization

Screen name	Purpose	Fields
<i>Attendant Console</i>	Define an attendant console.	All
<i>Data Module</i>	Define a data module.	All

Screen name	Purpose	Fields
<i>Feature Access Codes (FAC)</i>	Create FACs for the TTI merge and separate operations.	- Terminal Translation Initialization Merge Code - Terminal Translation Initialization Separation Code
<i>Feature-Related System Parameters</i>	Enable TTI for the system.	- TTI Enabled - TTI State - TTI Security Code
<i>Station</i>	Define a station.	All

Reports for Terminal Translation Initialization

The following reports provide information about the Terminal Translation Initialization feature:

- None

Considerations for Terminal Translation Initialization

This section provides information about how the Terminal Translation Initialization feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Terminal Translation Initialization under all conditions. The following considerations apply to Terminal Translation Initialization:

- None

Interactions for Terminal Translation Initialization

This section provides information about how the Terminal Translation Initialization feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Terminal Translation Initialization in any feature configuration.

- Attendant
You can separate an attendant if the attendant is in Position Available Mode. However, you cannot separate an attendant if any calls are in the queue, held, or active for the attendant.
- Attendant Night Service
You cannot separate the night service station, when the station is in night service.

- **Attendant Release Loop Operation**

If the attendant separates before the attendant-timed reminder-interval expires, the system reclassifies all calls that are held with the release loop operation by the attendant as attendant group calls.
- **Automatic Callback**

If you use TTI to separate either telephone that is part of Automatic Callback, the system breaks the automatic callback sequence.
- **Call Coverage**

Send All Calls and Goto Coverage remain active, while the telephone has no associated hardware. You can separate a telephone that is the receiver of Send All Calls or Goto Coverage. The system processes calls to the telephone as if the telephone is busy.
- **Call Coverage Answer Group**

If an extension that was an X port rejoins a call coverage answer group as a result of a TTI merge, the system excludes the extension from all transactions that are already active in the call coverage answer group.
- **Call Forwarding**

You can separate a telephone while Call Forwarding is active. If a destination extension for call forwarding separates, Call Forwarding to that extension remains active. The system processes calls to the telephone as if the telephone is busy.
- **Call Pickup**

If a line appearance is available, a member of a call pickup group can 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. The member can participate in all subsequent calls to the call pickup group.
- **Expert Agent Selection (EAS)**

Station user records cannot be shared between TTI ports and EAS login ID extensions. 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 and Direct Department Calling**

The system excludes telephones that are previously X-porting as a result of a TTI separation request from all transactions that are already active in the hunt group when the telephone is merged.
- **Site Data**

If Terminal Translation Initialization is enabled, the system displays a warning message after you administer the **Site Data** fields.

If Terminal Translation Initialization is enabled, and you change the **Port** field on the **Station** screen from **x** to a port number, and change the **Room**, **Jack**, or **Cable** fields in the **Site Data** section of the **Station** screen, the system displays a warning message when you tab off the fields.
- **Terminating Extension Group (TEG)**

If any member of the TEG, that was previously an X port as a result of TTI is merged, the member is excluded from all transactions that are already active in the TEG when the member is merged. The member can join all subsequent calls to the group.

Terminal Translation Initialization

Interactions for Terminal Translation Initialization

Terminating Extension Group

Use the Terminating Extension Group (TEG) feature to allow an incoming call to ring as many as four telephones at one time. Any user in the group can answer the call.

Detailed description of Terminating Extension Group

This section provides a detailed description of the Terminating Extension Group (TEG) feature.

You can administer any telephone as a TEG member. However, only a multiappearance telephone can be assigned a TEG button with a merged-status lamp. With the TEG button, the user can select a TEG call appearance to answer or bridge onto an existing call, but not to originate the call.

When a TEG members answers an incoming call, a temporary bridged appearance is maintained at the multiappearance telephones in the group. However, this appearance is not visible. Any TEG member can press the TEG button to bridge onto the call.

Hardware requirements for Terminating Extension Group

The Terminating Extension Group feature requires the following hardware:

- None

Administering Terminating Extension Group

This section describes the screens that you use to administer the Terminating Extension Group (TEG) feature

Screens for administering Terminating Extension Group

Screen name	Purpose	Fields
<i>Terminating Extension Group</i>	Define the groups for the TEG feature.	• Coverage Path • Group Name • Group Extension • COR • TN • ISDN Call Display • Ext

Reports for Terminating Extension Group

The following reports provide information about the Terminating Extension Group feature:

- None

Considerations for Terminating Extension Group

This section provides information about how the Terminating Extension Group (TEG) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Terminating Extension Group under all conditions. The following considerations apply to Terminating Extension Group:

- A telephone 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 system routes the additional calls accordingly.

Interactions for Terminating Extension Group

This section provides information about how the Terminating Extension Group (TEG) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Terminating Extension Group in any feature configuration.

- Automatic Callback
This feature cannot be active for a TEG.

- **Bridged Call Appearance**
Calls to a TEG cannot be bridged, except by way of 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 telephones. When a user presses Send Term, the system redirects calls to the TEG redirect to coverage. The merged status lamp lights on all telephones with a Send Term button. Any member with a Send Term button can press the button to deactivate Send Term. Incoming calls are directed to the group.
- **Call Park**
A TEG call cannot be parked on the group extension. However, group members who answer a call can park a TEG call on their own extensions.
- **Direct Department Calling (DDC) and Uniform Call Distribution (UCD)**
A TEG cannot be a member of a DDC or a UCD group.
- **Internal Automatic Answer**
TEG calls are not eligible for Internal Automatic Answer. However, calls that are placed to an individual extension are eligible.
- **Leave Word Calling (LWC)**
Leave Word Calling messages can be stored for a TEG. Any member of the group, covering user of the group, or system-wide message retriever can retrieve the stored messages. 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 telephones 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 telephones 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.

Transfer

Use the Transfer feature to allow telephone users to transfer trunk or internal calls to other telephones or trunks without attendant assistance.

Transfer supports the following capabilities:

- Pull Transfer
- Abort Transfer
- Transfer Recall
- Transfer Upon Hangup
- Trunk-to-Trunk Transfer
- Outgoing Trunk to Outgoing Trunk Transfer
- Emergency Transfer (Power Failure Transfer)

Detailed description of Transfer

This section provides a detailed description of the Transfer feature.

Pull Transfer

Use Pull Transfer to allow either the transferring or the 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, the parties are transferred with the standard transfer process.

To use Pull Transfer, calling parties and called parties must be on the same server, or called parties must be reached by way of Italian TGU/TGE tie trunks.

Called parties who use analog telephones flash the switch hook, or press the flash key or recall button to transfer calls. Called parties who use digital telephones press the transfer key to complete transfers.

Abort Transfer

Use Abort Transfer to stop 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 telephone 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

Transfer Recall

Use Transfer Recall to return the unanswered transfer call back to the person who transferred the call. Transfer Recall uses a priority alerting signal. The display on the telephone also shows *rt*, which indicates a returned call from a failed transfer operation.

Transfer Upon Hangup

Use Transfer Upon Hangup to hang up to transfer the call without having the need to press the Transfer button a second time. You press the Transfer button, dial the number the call is being transferred to and then hang up. Transfer Upon Hangup is an optional capability at the system level. You can still press the transfer button a second time to transfer the call.

Trunk-to-Trunk Transfer

With Trunk-to-Trunk Transfer, an attendant or a user can connect an incoming trunk call to an outgoing trunk.



SECURITY ALERT:

Trunk-to-trunk transfer poses a significant security risk. Use this capability with caution.

The system provides three levels of administration for this Trunk-to-Trunk Transfer:

- System-wide
- COR-to-COR
- 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*** screen. To allow individual users to control Trunk-to-Trunk Transfers, assign capabilities on the ***Class of Service*** screen.

Outgoing Trunk to Outgoing Trunk Transfer

With Outgoing Trunk to Outgoing Trunk Transfer (OTTOTT), a controlling party (such as a station user or an attendant), initiates two or more outgoing trunk calls, and then connects the trunks. This operation removes the controlling party from the connection, and conferences the outgoing trunks. The controlling party can also establish a conference call with the outgoing trunks, and drop out of the conference, and leave only the outgoing trunks on the conference.

NOTE:

This capability is an optional enhancement to Trunk-to-Trunk Transfer and requires careful administration and use. The Distributed Communications System (DCS) Trunk Turnaround may be an acceptable and safer alternative to this feature.

With OTTOTT, you can establish calls in which the only parties involved are external to Avaya Communication Manager and are on outgoing trunks. This type of call can result in locked-up trunks, such as trunks that cannot be disconnected except by busy-ing-out and releasing the affected trunk circuit.

To clear the lockup, a service technician must reseal the trunk board, or busy-out and release the affected trunk.

Emergency Transfer

Emergency Transfer provides service to and from the central office (CO) to the local telephone company during a power failure, or when service is impaired. Emergency Transfer is also called Power Failure Transfer.

With Emergency Transfer, users of the 500- or 2500-type analog telephones can access the local CO and answer incoming calls during a power failure. Each server cabinet supports Emergency Transfer panels by way of the AUX connectors on the rear panel. The transfer is initiated when:

- A transfer panel or the 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. These telephones can either be pulse-dialing or touch-tone phones. You must use pulse dialing if the CO accepts only dial pulses. Each telephone can be connected to a separate CO.

When the system is not in Emergency Transfer mode, you can use the transfer telephones as regular telephones.

Hardware requirements for Transfer

The Transfer feature requires the following hardware:

- None

Administering Transfer

This section describes the screens that you use to administer the Transfer feature.

Screens for administering Transfer

Screen name	Purpose	Fields
<i>Feature-Related System Parameters</i>	Define Transfer options	

Reports for Transfer

The following reports provide information about the Transfer feature:

- None

Considerations for Transfer

This section provides information about how the Transfer feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Transfer under all conditions. The following considerations apply to Transfer:

- You can administer transferred trunk calls to receive either music or silence if the first part of the transfer places the call on hold.
- Multiappearance telephones must have an idle appearance to transfer a call.
- Single-line telephone users momentarily flash the switch hook 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 telephone transfers the call.

- Users of Digital Communications Protocol (DCP), hybrid, and wireless telephones can transfer a call that is 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. Communication Manager assumes the transfer is for the call on hold, and the transfer feature works as usual.

If more than one call is 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, Communication Manager 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.

- Communication Manager 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 Integrated Services Digital Network-Basic Rate Interface (ISDN-BRI) display phones. All of these telephones, except for the Hybrids, can display the confirmation message in English, Spanish, French, Italian, or a language you define. Hybrid telephones 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.

Transfer - Outgoing Trunk to Outgoing Trunk

- 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.

Transfer - Trunk to Trunk

- 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 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 for Transfer

This section provides information about how the Transfer feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Transfer in any feature configuration.

- 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 by way of Pull Transfer.

- BRI telephones

Callers who use BRI Stations reach desired parties through the intermediate step of 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 software 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 by way of 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.

Outgoing Trunk to Outgoing Trunk Transfer (OTTOTT)

- **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. If any remaining party is an answered station or attendant, accept turnaround.
 - b** 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 by way of 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.
 - c** If any remaining party is an outgoing trunk administered for OTTOTT that has received answer supervision, accept turnaround.
 - d** If any remaining party is an outgoing DCS trunk, forward the turnaround request.
 - e** Otherwise, deny turnaround.
- **Incoming Disconnect Supervision**
Outside of the U.S., incoming disconnect supervision is a switching 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 (PCOLs)**
Transfer of 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 (RLT)

RLTs are used by 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 by way of 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, you use **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.

Trunk to Trunk Transfer

- 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 vector directory number (VDN) that ultimately routes to a destination off-net.

A route to a number off-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.

Emergency Transfer

- Night Service

If a power failure occurs when the system is in night service, the system automatically returns to night service when power is restored.

Trunk Flash

Use the Trunk Flash feature to allow a user of a multifunction telephone or an attendant console to access far-end customized services or central office (CO) customized services.

Detailed description of Trunk Flash

This section provides a detailed description of the Trunk Flash feature.

With Trunk Flash, a user of a multifunction telephone or an attendant console can gain access to far-end customized services or central office (CO) customized services. To use Flash Trunk, the user presses the flash button or enters the feature access code (FAC) for Trunk Flash.

CO customized services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from a Communication Manager telephone on an active trunk call. The Trunk Flash feature can help to reduce the number of trunk lines that are connected to the server that runs Communication Manager by:

- Perform trunk-to-trunk call transfers at the far-end or the 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.
- Perform 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 multifrequency (DTMF) telephone sets used in Italy and the United Kingdom are equipped with a Flash button. When a user presses that flash button, the button generates a rotary digit 1. When an analog telephone that is administered as a DTMF telephone, for example, as a 2500 or a 71nn-type telephone, transmits a rotary digit 1, the system processes the signal as a recall signal from the telephone set to Communication Manager.

A centralized attendant service (CAS) attendant who is connected to a release line trunk (RLT), the flash button controls certain CAS features at the branch. For a user of a multifunction telephone or non-CAS attendant connected to a central office CO, foreign exchange (FX), or Wide Area Communications 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 two-way call directions on selected 2-wire analog 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 the CO that provides the customized services, use of the Trunk Flash signal can cause the call to be disconnected by the far end or CO.

The system does not record Call Detail Recording (CD) information for calls that a user makes with the Trunk Flash feature.

Hardware requirements for Trunk Flash

The Trunk Flash feature requires the following hardware:

- None

Administering Trunk Flash

This section describes the screens that you use to administer the Trunk Flash feature.

Screens for administering Trunk Flash

Screen name	Purpose	Fields
<i>Feature Access Code (FAC)</i>	Assign an FAC for Trunk Flash.	Flash Access Code
<i>Trunk Group</i>	Enable Trunk Flash for the trunk group.	Trunk Flash

End-user procedures for Trunk Flash

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Trunk Flash:

- Using Trunk Flash

To use Trunk Flash, perform one of the following actions:

- Press the Flash button.
- Enter the feature access code for Trunk flash.

Reports for Trunk Flash

The following reports provide information about the Trunk Flash feature:

- None

Considerations for Trunk Flash

This section provides information about how the Trunk Flash feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Trunk Flash under all conditions. The following considerations apply to Trunk Flash:



CAUTION:

The Trunk Flash feature allows the telephone user to receive central office (CO) dial tone. A user can place a call that is not monitored by Communication Manager, and is not subject to restrictions such as toll, facilities restriction level (FRL), and class of restriction (COR). Use caution when you enable this feature.

- 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 by way of 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.
- System features, such as internal conference call, transfer, and call par, can be combined with custom services. The custom services are CO-based features that are activated/controlled by sending a flash signal over the trunk to the CO. However, mixing Communication Manager features with custom services causes complications for the user when the user tracks a call. Communication Manager cannot give the local telephone user status information on the custom services.
- The Trunk Flash feature can only be accessed if the call has only one trunk. The trunk must be an outgoing trunk and the **Trunk Flash** field on the **Trunk Group** screen must be set to **y**. The Trunk Flash feature is disabled when the call involves more than one trunk, even if all the trunks have Trunk Flash enabled.
- The system allows five users to participate in a conference call with the trunk line party. However, to access the Trunk Flash feature, at least one of the user telephones must have a Flash button.
- With a call that involves more than one user, one of the users can press the Flash button, and another user can dial the telephone number. The user that dials the telephone number is not required to have a telephone with a Flash button.

If the far end or central office (CO) does not support custom services, the far end or CO can ignore the call or drop the call. If the far end or CO ignores or drops the call, the user might hear a clicking sound or the user might hear silence.

Interactions for Trunk Flash

This section provides information about how the Trunk Flash feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Trunk Flash in any feature configuration.

- None

Trunk Flash

Interactions for Trunk Flash

Uniform Dial Plan

Use the Uniform Dial Plan (UDP) capability to share a common dial plan among a group of servers. The UDP applies both interserver dialing and intraserver dialing. The UDP provides the following types of dial plans:

- 3-digi
- 4-digit
- 5-digit
- 6-digit
- 7-digit
- A combination of any of these dial plans

UDP provides extension-to-extension dialing among two or more private-switching systems.

You can use UDP with the following entities:

- Main servers
- Tributary servers
- Satellite servers
- Electronic Tandem Networks (ETN)
- Distributed Communication Systems (DCS)

Note that you must use a 4-digit dial plan or a 5-digit dial plan for DCS.

Detailed description of Uniform Dial Plan

This section provides a detailed description the Uniform Dial Plan (UDP) capability.

The software uses the UDP to route a call off the local server. The user dials an extensions that consists of 3 to 7 digits. The software converts the extension that the user dials into a private-network number or a public-network number.

To convert the extension, the software substitutes digits at the front of the extension that the user dials. The software supports the following types of extension number conversions:

- Automatic Alternate Routing (AAR)

The system uses AAR routing information to route calls within your company over your own private network.

The software converts the number that the user dials. The software then analyzes the number, and routes the call as a private-network call.

- Automatic Route Selection (ARS)

The system uses ARS routing information to route calls that go outside your company over public networks. The system also uses ARS routing information to route calls to remote company locations if you do not have a private network.

The software converts the number that the user dials. The software then analyzes the number, and routes the call as a public-network call.

- Extension number portability (ENP)

If you want calls on your system to use ENP conversion, you must specify a node number. The software uses the node number to determine the routing pattern for the call. If the user dials an extension that consists of 4 to 6 digits, the software chooses an ENP code that is based on the first one or two digits of the extension. The software does not use the ENP code for routing. Therefore the ENP code is independent of location.

- Extension (EXT)

The system uses EXT conversion to analyze the extension that the user dials as an extension.

Unlike AAR conversion or ARS conversion, the system might not change the extension number that the user dials, before the system routes the call. If no a UDP entry for a particular extension exists, or for a range of extensions, the system considers the extensions to be local extension.

You specify a UDP for individual extensions or groups of extensions that have the same leading digits. For example, if you use a 5-digit UDP, and choose a matching pattern of “123,” all 5-digit extensions that begin with “123” have the same UDP conversion scheme. If you use a 5-digit UDP, and want the software to use the UDP for only one extension, you must create a matching pattern that is the same as the extension. For example, if you choose a 5-digit UDP for extension 12345, you must specify 12345 as the matching pattern.

Each user extension can be assigned to one of the following six treatments:

- UDP Code
 - Conversion to AAR with a given location code
 - Further conversion is suppressed
- AAR Code
 - Conversion to AAR with a given location code
 - Further conversion is suppressed
- ENP Code
 - Conversion to a private-network number
 - Route to the given node number routing
- Temp OOS
 - Temporarily out of service
 - Give reorder
- Local
- Blank

Similar to local, but this extension is not chosen when you run the add station command.

When a user dials an extension that is on a server that is included in a UDP, the software firsts determines if the extension is assigned to a local station on the server. If the extension is assigned to a local station on the server, the software routes the call to the station. The software does not convert the extension numbers.

When a user dials an extension that is not assigned to a local station on the server, the software compares the extension with the matching patterns. If the software finds a match between the extension and the matching patterns, the software converts the extension into a private network number. The software then routes the call as specified by the conversion.

When a user dials an extension, and the extension matches more than one matching pattern, the software selects the pattern that has the most matching digits. The software compares the extension and the matching pattern starting with the first digits that the user dials through to the last digits. See [Table 95, Matching pattern selections](#), on page 1111, for examples of matching patterns.

If the extension does not match a matching pattern, and Extended Trunk Access (ETA) is enabled on the system, the software uses ETA to route the call. If the extension does not match a matching pattern, and ETA is not enabled on the system, the user receives intercept treatment.

Table 95: Matching pattern selections

Extension dialed	Matching patterns available	Matching pattern selected
123	123 and 1234	123
12345	123 and 1234	1234
12355	123 and 1234	123

Uniform Dial Plan example

The section contains an example of UDP administration, and the processing of several calls that use UDP.

To administer UDP, you must 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 the system routes a UDP call.
- As either local or remote to Communication Manager.

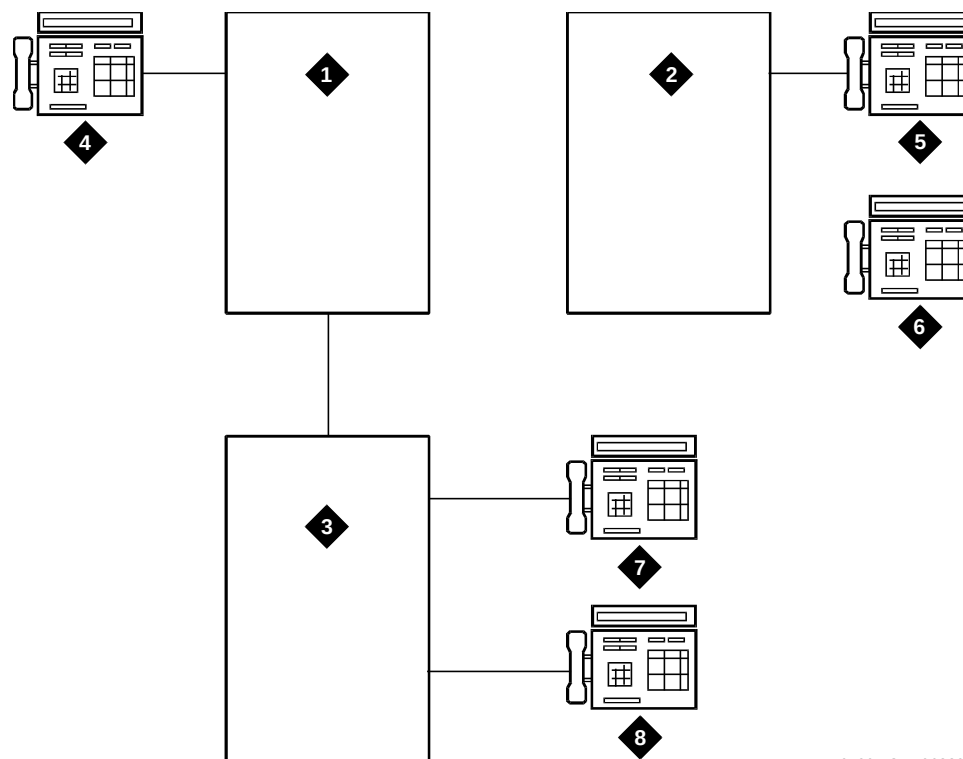
The same 5-digit extension is used to call a station, regardless of where in the ETN that call originates. This example includes three media servers or switches ([Table 96, RNX and UDP codes](#), on page 1111). Each media server and switch has a list of RNX and UDP codes.

Table 96: RNX and UDP codes

Media server or switch	RNX	UDP code
A	224	41
C	223	51
C	223	52
B	222	60
B	222	61

See [Figure 292, UDP Example](#), on page 1112 for an example of a UDP scenario.

Figure 292: UDP Example



r764905 C JL 062296

Figure notes

1	Switch A The dial plan for the extensions is 41, RNX=224.	4	Extension 41000
2	Switch B The dial plan for the extensions is 60 and 61, RNX=222.	5	Extension 61234
3	Switch C The dial plan for the extensions is 51 and 52, RNX=223	6	Extension 60123
		7	Extension 51234
		8	Extension 5200

A user at extension 41000 who wants to call extension 61234, has one of the following options:

- Dial 61234
- Dial the AAR access code, and then dial 222-1234

If the user dials 61234, the system:

- Recognizes 61 as a remote UDP
- Determines that the associated RNX is 222
- Uses AAR to route the call to 222-1234

If the user dials the AAR access code and 222-1234, the system:

- Finds the route pattern for RNX 222
- Routes the call to the server associated with that RNX

When the software uses UDP to route a call to another server or switch, the route pattern provides the correct digit deletion and insertion instructions, so the receiving switch gets numbers in the format that the receiving switch expects. You can configure the software several different ways.

- If AAR is available on the receiving media server or switch:
 - Subnetwork trunking can be used to insert the FAC for AAR on the server or switch where the call originates
 - Digit insertion can be used to insert the FAC for AAR on the receiving server

The receiving server uses AAR digit conversion to delete 3 digits and add the digit 6, thus converting the number 222 with 7 digits to an extension.

- If AAR is not available on the receiving media server or switch, subnet trunking must be used on the originating server or switch to delete the 222 and insert the digit 6 at the start of the extension number. This conversion ensures that the receiving server can continue to route the call correctly.

If the user at extension 51234 on media server C dials extension 61234, the call must first go through media server A before proceeding to media server B. When the user dials 61234:

- The software recognizes 61 as a UDP code.
- The software determines that the associated RNX is 222, and uses the AAR feature to route the call.
- The AAR feature access code plus 222-1234 are outputted to media server A.
- Media server A then recognizes the RNX 222 as a remote server or switch, and routes the call to media server B and extension 61234.

This same type of call routing occurs when an extension at media server B calls an extension at media server C.

If extension 61234 on media server B calls extension 61235, the software recognizes 61 as a local UDP code, and the system routes the call directly to extension 61235.

Hardware requirements for Uniform Dial Plan

The Uniform Dial Plan capability requires the following hardware:

- None

Administering Uniform Dial Plan

The following steps are part of the administration process for the Uniform Dial Plan (UDP) capability:

- [Administering the Uniform Dial Plan table](#) on page 1116
- [Administering the route pattern](#) on page 1117
- [Administering the Node Number Routing table](#) on page 1126
- [AAR Digit Conversion Table screen](#) on page 1127
- [Administering the ARS Digit Conversion table](#) on page 1128
- [Administering the AAR Digit Analysis table](#) on page 1129
- [Administering the ARS Digit Analysis table](#) on page 1132
- [Administering the extension number portability numbering plan](#) on page 1134

This section describes:

- Any prerequisites for administering the Uniform Dial Plan capability
- The screens that you use to administer the Uniform Dial Plan capability
- Complete administration procedures for the Uniform Dial Plan capability

Prerequisites for administering Uniform Dial Plan

You must complete the following actions before you can administer the Uniform Dial Plan (UDP) capability:

- Ensure that the Uniform Dial Plan capability is enabled on your system.
- Ensure that a feature access code (FAC) for automatic alternate routing (AAR) and an FAC for automatic route selection (ARS) are available on your system

To ensure that the Uniform Dial Plan capability is enabled on your system:

- On the **Optional Features** screen, ensure that the **Uniform Dialing Plan** field is set to **y**. To view the screen, type **display system-parameters customer-options**. Press **Enter**. If the **Uniform Dialing Plan** field is set to **n**, your system is not enabled for the Uniform Dial Plan capability. Contact your Avaya representative for assistance before continuing with this procedure.

For a complete description of the **Optional Features** screen, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

To ensure that an FAC for AAR and ARS are available on your system:

- 1 Type **change feature-access-codes**. Press **Enter**.

The system displays the **Feature Access Code (FAC)** screen ([Figure 293, Feature Access Code \(FAC\) screen](#), on page 1115).

Figure 293: Feature Access Code (FAC) screen

FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1	Access Code:	*00
Abbreviated Dialing List2	Access Code:	*01
Abbreviated Dialing List3	Access Code:	*02
Abbreviated Dial - Prgm Group List	Access Code:	*03
Announcement	Access Code:	*04
Answer Back	Access Code:	*05
Auto Alternate Routing (AAR)	Access Code:	8
Auto Route Selection (ARS) -	Access Code 1:	*9
	Access Code 2:	9
Automatic Callback	Activation:	*06
	Deactivation:	#06
Call Forwarding Activation Busy/DA:	*07	All: *08
	Deactivation:	#09
Call Park	Access Code:	*10
Call Pickup	Access Code:	*11
CAS Remote Hold/Answer Hold-Unhold	Access Code:	*50
CDR Account Code	Access Code:	*12
Change COR	Access Code:	
Change Coverage	Access Code:	*13
Contact Closure	Open Code:	
Contact Closure	Pulse Code:	
	Close Code:	

- 2 Page through the screens until you see the Auto Alternate Routing (AAR) Access Code and the Auto Route Selection (ARS) fields.
- 3 Perform one of the following actions:
 - If the Auto Alternate Routing (AAR) Access Code and the Auto Route Selection (ARS) fields each contain an FAC, press **Cancel**.
 - If the Auto Alternate Routing (AAR) Access Code field and the **Auto** Route Selection (ARS) do not contain an FAC, type an FAC in the fields and press **Enter** to save your changes.

For more information on the Feature Access Code feature, including information on how to change or deactivate an FAC, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Uniform Dial Plan

Screen name	Purpose	Fields
<i>Uniform Dial Plan Table</i>	Define the type of conversion, the matching pattern, and the insertion digits.	All
<i>AAR Digit Analysis Table</i>	Define automatic alternate routing (AAR) digit analysis.	All
<i>ARS Digit Analysis Table</i>	Define automatic route selection (ARS) digit analysis.	All
<i>AAR Digit Conversion Table</i>	Define AAR digit conversion.	All
<i>ARS Digit Conversion Table</i>	Define ARS digit conversion.	All

Screen name	Purpose	Fields
Extension Number Portability (ENP) Numbering Plan	Define ENP codes.	All
Number Node Routing	Associate a route pattern with each node in the ENP subnetwork.	Route Pat
Route Pattern	Define the route pattern.	All
Optional Features	Ensure that the Uniform Dial Plan capability is enabled on your system.	Uniform Dialing Plan

Administering the Uniform Dial Plan table

To administer the Uniform Dial Plan Table for your system:

- 1 Type **change uniform-dial plan *n***, where *n* is the number of the dial plan that you want to administer. Press **Enter**.

The system displays the **Uniform Dial Plan Table** screen ([Figure 294, Uniform Dial Plan Table screen](#), on page 1116).

Figure 294: Uniform Dial Plan Table screen

UNIFORM DIAL PLAN TABLE													
													Percent Full: 0
Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num	Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
21	5	1	255	aar	y							n	
25	5	1	364	aar	y							n	
5	5	1	510	aar	n							n	
51	5	1	515	aar	n							n	
6	5	1	202	aar	n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	
					n							n	

- 2 In the **Matching Pattern** field, type the number that you want the software to match to the number that a user dials. You can type digits 0 through 9. You can type a maximum of seven digits.
- 3 In the **Len** field, type the number of digits of the user-dialed number, that the software analyses when the software compares the user-dialed number with the numbers in the matching pattern field. You can type digits 3 through 7.
- 4 In the **Del** field, type the number of digits that the software deletes before the software routes a call. You can type digits 0 through 3. The number that you type must be less than, or equal to, the number that you type in the **Len** field.
- 5 In the **Insert Digits** field, type the digits that the software uses to replace the digits that the software deletes from the user-dialed number. If you want the software to delete the digits rather than replace the digits, leave the **Insert Digits** field blank. You can type digits 0 through 9. You can type a maximum of 4 digits.
- 6 In the **Net** field, type the server or switch network that the software uses to analyze the number that the software converts. Perform one of the following actions:
 - Type **ext** if you want the software to route the converted digit-string as the converted digit-string as an extension.
 - Type **aar** if you want the software to route the converted digit-string as the converted digit-string as its converted AAR address.
 - Type **ars** if you want the software to route the converted digit-string as its converted ARS address.
 - Type **enp** if you want the software to route the converted digit-string as its ENP node number.If you type **enp**, you must:
 - Type the ENP (extension number portability) node number in the **Node Number** field.
 - Leave the **Insert Digits** field blank.
 - Type **n** in the **Conv** field.
- 7 In the **Node Num** field, type the ENP node number. You can type digits 1 through 999.
- 8 In the **Conv** field, perform one of the following actions:
 - Type **y** if you want additional digit conversion.
 - Type **n** if you do not want additional digit conversion.
- 9 The **Percent Full** field displays the percent of the allocated uniform dial plan data resources that are currently used. You cannot change this field.
- 10 Press **Enter** to save your changes.

Administering the route pattern

To administer the route pattern for your system:

- 1 Type **change route-pattern *n***, where ***n*** is the number of the route pattern. Press **Enter**.
The system displays the **Route Pattern** screen ([Figure 295, Route Pattern screen](#), on page 1118).

The **Route Pattern** screen defines the route patterns used by your server. 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, the system can insert the dial access code for an alternative carrier into the digit string.

Figure 295: Route Pattern screen

Pattern Number: 1													Pattern Name:	
Secure SIP? n														
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted							
No			Mrk	Lmt	List	Del	Dgts							
1:	1	0	303			0	9							
2:														
3:														
4:														
5:														
6:														
BCC	VALUE	TSC	CA-TSC			ITC	BCIE	Service/Feature	BAND	No.	Numbering	LAR		
0	1	2	3	4	W	Request				Dgts	Format			
1:	y	y	y	y	n	n						rest		
2:	y	y	y	y	n	n						rest		
3:	y	y	y	y	n	n						rest		
4:	y	y	y	y	n	n						rest		
5:	y	y	y	y	n	n						rest		
6:	y	y	y	y	n	n						rest		
													Subaddress	

- 2 In the **Band** field, 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.

This field appears when the **Services/Features** field is **outwats-bnd** and when either the **ISDN-PRI** or **ISDN-BRI Trunks** field is set to **y** on the **Optional Features** screen. The **Band** field is required by Call-by-Call Service Selection.

- 3 In the **BCC Value** field, type information in any of the BCC Value columns.

- In the **0** column, perform one of the following actions:
 - a Type **y** if the BCC is appropriate for the associated route pattern.
 - b Type **n** if the BCC is not appropriate for the associated route pattern.
- In the **1** column, perform one of the following actions:
 - a Type **y** if the BCC is appropriate for the associated route pattern.
 - b Type **n** if the BCC is not appropriate for the associated route pattern.
- In the **2** column, perform one of the following actions:
 - a Type **y** if the BCC is appropriate for the associated route pattern.
 - b Type **n** if the BCC is not appropriate for the associated route pattern.

- In the **3** column, perform one of the following actions:
 - a** Type **y** if the BCC is appropriate for the associated route pattern.
 - b** Type **n** if the BCC is not appropriate for the associated route pattern.
- In the **4** column, perform one of the following actions:
 - a** Type **y** if the BCC is appropriate for the associated route pattern.
 - b** Type **n** if the BCC is not appropriate for the associated route pattern.
- In the **W** column, perform one of the following actions:
 - a** Type **y** if the BCC is appropriate for the associated route pattern.
 - b** Type **n** if the BCC is not appropriate for the associated route pattern.

The **BCC Value** field (Bearer Capability Class) identifies the type of call appropriate for this trunk group, such as voice calls and different types of data calls. This field appears when either the **ISDN-PRI** or **ISDN-BRI Trunks** field is set to **y** on the **Optional Features** screen.

See [Table 97, BCC Value field entries](#), on page 1119 for a description of BCC Values.

Table 97: BCC Value field entries

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)

- 4 In the **BCIE** field, type the value that determines the creation of the ITC codepoint in the setup message. The **BCIE** field applies to ISDN trunks. You can type **ept** for endpoint or **unr** for unrestricted.

The software displays the **BCIE** field when the **ITC** field is set to **both**.

- 5 In the **CA-TSC** field, type the information for ISDN B-channel connections.

See [Table 98, CA-TSC field entries](#), on page 1119 for a description of the entries for the **CA-TSC** field.

Table 98: CA-TSC field entries

CA-TSC entries	Usage
as-needed	The CA-TSC is set up only when needed. This causes a slight delay. Avaya recommends this entry 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.

- 6 In the DCS/QSIG Intw, field perform one of the following actions:

- Type **y** to enable CS/QSIG Voice Mail Interworking.
- Type **n** to disable CS/QSIG Voice Mail Interworking.

The DCS/QSIG Intw field appears when the Interworking with DCS field on the **Optional Features** screen is set to **y**.

- 7 In the FRL field, enter the Facility Restriction Level (FRL) associated with the preference.

You can type the numbers **0** through **7**.

0 is the least restrictive FRL and 7 is the most restrictive FRL. The FRL of the calling party must be greater than, or equal to, this FRL to access the associated trunk-group.



SECURITY ALERT:

For system security reasons, Avaya recommends that you use the most restrictive FRL possible.

- 8 In the Grp No field, type the trunk group number associated with the preference.

You can type the numbers **1** through **666** for DEFINITY R, CSI, and SI.

You can type the numbers **1** through **2000** for the S8300 Media Server, the S8700 IP-Connect, and the S8700 Multi-Connect.

- 9 In the Hop Lmt field, type the number of hops for each preference. A *hop* occurs when a call tandems through a media server or switch to another destination. You limit the number of hops to prevent circular hunting, which ties up trunk facilities without completing a call. The software blocks a hop equal to or greater than the number you enter in the Hop Lmt field.

See [Table 99, Hop Lmt field entries](#), on page 1120 for the Hop Lmt field entries.

Table 99: Hop Lmt field entries

Hop Lmt 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.

- 10 In the Inserted Digits field, type the digits that the software inserts for routing.

You can type the digits **0** through **9**. You can type a maximum of 36 digits.

The software can send a maximum of 52 digits. This includes the digits that you enter in the Inserted Digits field, plus the digits that the user dials. The software counts each special symbol as two digits.

See [Table 100, Inserted Digits field entries](#), on page 1121 for the Inserted Digits field entries.

Table 100: Inserted Digits field entries

Inserted Digits entries	Usage
*	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.
+	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 <i>Feature-Related System Parameters</i> 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)
p	The associated trunk group must be of type “sip.” Enter the single digit p for fully qualified E.164 numbers. The p is translated to a + and is placed at the front of the digit string.

- 11** In the **ITC** field, type the 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.

The ITC field must be set to either **unre** or **both**, if the **w** column of the BCC field is set to **y**.

See [Table 101, ITC field entries](#), on page 1121 for **ITC** field entries.

Table 101: ITC field entries

ITC entries	Usage
both	Calls from restricted and unrestricted endpoints can access the route pattern.
rest	Calls from restricted endpoints can access the route pattern.
unre	Calls from unrestricted endpoints can access the route pattern.

- 12** In the **IXC** field, type the Inter-Exchange Carrier (IXC) carrier, such as AT&T. The IXC information is used for calls that route via an IXC and for Call Detail Recording (CDR). The system displays the IXC field when the **ISDN - PRI** field or **ISDN-BRI Trunks** field on the *Optional Features* screen is set to **y**.

See [Table 102, IXC field entries](#), on page 1122 for **IXC** field entries.

Table 102: IXC field entries

IXC entries	Usage
Valid carrier code	Identifies the carrier for IXC calls
user	For pre subscribed carrier. Used when an IXC is not specified.
none	The IXC 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.

- 13** In the LAR field, type the routing-preference for Look Ahead Routing.
See [Table 103, LAR field entries](#), on page 1122 for LAR field entries.

Table 103: LAR field entries

LAR 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.

- 14** In the **No. Del. Digits** field, type total number of digits you want the system to delete before it sends the number out on the trunk.

You can type the numbers **0** through **28**. You can make the field **blank**. To make the field blank, press **Clear**.

The software uses this information to modify the dialed number so an AAR or ARS call routes over different trunk groups that terminate in media servers or switches with different dial plans. The software uses this information for the calls that use the following routing methods:
- To or through a remote media server or switch
 - Over tie trunks to a private network server or switch
 - Over Central Office (CO) trunks to the serving CO
- 15** In the **No. Dgts Subaddress** field, type the number of dialed digits to send in the calling party subaddress IE.

You can change the **No. Dgts Subaddress** field if the **ISDN Feature Plus** field on the **Optional Features** screen is set to **y**.

You can type the numbers **1** through **5**. You can make the field **blank**. To make the field blank, press **Clear**.

The software uses this information to route a call to a number where the media server deletes the dialed number and inserts the listed directory number (LDN). The LDN is then 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 to the attendant.

- 16** In the **NPA** field, type the 3-digit Numbering Plan Area (NPA) (or area code) for the terminating endpoint of the trunk group.

You can type the digits **0** through **9**. You can make the field **blank**. To make the field blank, press **Clear**.

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.

Call your local telephone company to verify the NPA, if you need help.

You do not need to enter an NPA for AAR.

- 17** In the **Numbering Format** field, type the information that specifies the format of the routing number used for the ISDN trunk groups for this preference.

Note that, to access the Bellcore NI-2 Operator Service Access, you must type **unk-unk** in the **Inserted Digits**.

The **Inserted Digits** field appears when the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y** on the **Optional Features** screen.

See [Table 104, Numbering Format field entries](#), on page 1123 for **Numbering Format** field entries.

Table 104: Numbering Format field entries

Numbering Format 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)
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)

- 18** The **Pattern Number** field, is a display-only field that displays the route pattern number. The route pattern number is from **1** to **640**.

- 19** In the **Prefix Mark** field, type the prefix mark information for ARS.

You can type the numbers **0** through **4**.

You can make the field **blank**. To make the field blank, press **Clear**.

This entry is not required for AAR.

The prefix marks set the requirements for sending a prefix digit 1, indicating a long-distance call. Prefix marks apply to 7-digit Direct Distance Dialing (DDD) or 10-digit DDD public network calls. A prefix digit 1 is sent only when **call type** is a foreign number plan area (FNPA) or home numbering plan area (HNPA) in the **ARS Digit Analysis Table** screen.

For a WATS trunk, the prefix mark is the same as the local CO trunk.

See [Table 105, Prefix Mark field entries](#), on page 1124 for **Prefix Mark** field entries.

Table 105: Prefix Mark field entries

Prefix Mark 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 media server or a switch that rejects calls with a prefix digit 1.
blank	For tie trunks, leave this field blank.

Service/Feature

20 In the **Service/Feature** field, type the information element (IE) in a call in this route pattern.

You can type from 1 to 15 characters,

The **Service/Feature** field is required by Call-by-Call Service Selection, and Network Call Redirection and Transfer.

The system displays the **Service/Feature** field when the **ISDN - PRI** field or the **ISDN - BRI Trunks** field on the **Optional Features** screen is set to **y**.

See [Table 106, Service/Feature field entries](#), on page 1125 for **Service/Feature** field entries.

Table 106: Service/Feature field entries

Service Feature entries	Description
accunet	—
i800	—
inwats	—
lds	—
mega800	—
megacom	—
multiquest	—
operator	—
oper-lds	Operator and lds
oper-meg	Operator and megacom
oper-sdn	Operator and sdn
outwats-bnd	—
sdn	Enter to allow Network Call Redirection/Transfer.
sub-operator	—
sub-op-lds	Sub-operator and lds
sub-op-meg	Sub-operator and megacom
sub-op-sdn	Sub-operator and sdn
wats-max-bnd	—

- 21** In the **Toll List** field, type 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 set to **2** or **3**.

You can type the numbers 1 through 32.

You can make the field **blank**. To make the field blank, press **Clear**.

This entry is not required for AAR.TSC

- 22** In the **TSC** field, perform one of the following actions:

- Type **y**:
 - To allow Call-Associated TSCs, and to allow incoming Non-Call-Associated TSC requests to be tandemed out for each preference
 - For feature transparency on DCS+ calls and to use QSIG Call Completion
- Type **n** if you do *not* want:
 - To allow Call-Associated TSCs, and to allow incoming Non-Call-Associated TSC requests to be tandemed out for each preference
 - Feature transparency on DCS+ calls and to use QSIG Call Completion.

Administering the Node Number Routing table

To administer node number routing for your system:

- 1 Type **change node-number routing *n***, where *n* is the number of the node. Press **Enter**.

The system displays the *Node Number Routing* screen ([Figure 296, Node Number Routing screen](#), on page 1126).

Figure 296: Node Number Routing screen

NODE NUMBER ROUTING						
Partitioned Group Number: 1						
Route Pat	Route Pat	Route Pat	Route Pat	Route Pat	Route Pat	Route Pat
1:	15:	30:	45:	60:	75:	90:
2:	16:	31:	46:	61:	76:	91:
3:	17:	32:	47:	62:	77:	92:
4:	18:	33:	48:	63:	78:	93:
5:	19:	34:	49:	64:	79:	94:
6:	20:	35:	50:	65:	80:	95:
7:	21:	36:	51:	66:	81:	96:
8:	22:	37:	52:	67:	82:	97:
9:	23:	38:	53:	68:	83:	98:
10:	24:	39:	54:	69:	84:	99:
11:	25:	40:	55:	70:	85:	
12:	26:	41:	56:	71:	86:	
13:	27:	42:	57:	72:	87:	
14:	28:	43:	58:	73:	88:	
	29:	44:	59:	74:	89:	

- 2 In the **Route Pat** field, type the number of the routing pattern that you want to use.
- 3 Press **Enter** to save your changes.

Administering the AAR Digit Conversion table

To administer the AAR Digit Conversion Table for your system:

- 1 Type **change aar digit-conversion *n***, where *n* is the number of the AAR Digit Conversion table. Press **Enter**.

The system displays the *AAR Digit Conversion Table* screen ([Figure 297, AAR Digit Conversion Table screen](#), on page 1127).

Note that the system sorts the screen information by matching pattern. The software displays numbers first, followed by characters in alphabetical order.

Figure 297: AAR Digit Conversion Table screen

[illegible]

- 2 In the **Matching Pattern** field, type the number that you want the software to match to the numbers that the user dials. If you require a prefix digit of 1 for 10-digit direct distance dialling (DDD), start the matching pattern with a 1. You can type:
 - Digits 0 through 9
 - A maximum of 18 digits
 - * or x or X as wild cards
- 3 In the **Min** field, type the minimum number of user-dialed digits that the software uses to compare with the matching pattern in the **Matching Pattern** field. You can type 1 to the number in the **Max** field.
- 4 In the **Max** field, type the maximum number of user-dialed digits the software uses to compare with the matching pattern in the **Matching Pattern** field. You can type from the number in the **Min** field to 28.
- 5 In the **Del** field, type the number of digits that you want the software to delete from the beginning of the dialed string. You can type 0 to the number in the **Min** field.
- 6 In the **Net** field, type the call-processing server network that the system uses to analyze the converted number.
- 7 In the **Conv** field, perform one of the following actions:
 - Type y if you want additional digit conversion.
 - Type n if you do not want additional digit conversion.
- 8 Use the **ANI Req** field only if the **Request Incoming ANI (non-AAR/ARS)** field on the **Mutifrequency-Signaling-Related System Parameters** screen is set to n. If **Request Incoming ANI (non-AAR/ARS)** field on the **Mutifrequency-Signaling-Related System Parameters** screen is set to n, perform one of the following actions:
 - Type y in the **ANI Req** field if you require ANI on incoming R2-MFC or Russian MF ANI calls.
 - Type n in the **ANI Req** field if you do not require ANI on incoming R2-MFC or Russian MF ANI calls.

- 4 In the **Max** field, type the maximum number of user-dialed digits the software uses to compare with the matching pattern in the **Matching Pattern** field. You can type from the number in the **Min** field to **28**.
- 5 In the **Del** field, type the number of digits that you want the software to delete from the beginning of the dialed string. You can type **0** to the number in the **Min** field.
- 6 In the **Net** field, type the call-processing server network that the system uses to analyze the converted number.
- 7 In the **Conv** field, perform one of the following actions:
 - Type **y** if you want additional digit conversion.
 - Type **n** if you do not want additional digit conversion.
- 8 Use the **ANI Req** field only if the **Request Incoming ANI (non-AAR/ARS)** field on the **Mutifrequency-Signaling-Related System Parameters** screen is set to **n**. If the **Request Incoming ANI (non-AAR/ARS)** field on the **Mutifrequency-Signaling-Related System Parameters** screen is set to **n**, perform one of the following actions:
 - Type **y** in the **ANI Req** field if you require ANI on incoming R2-MFC or Russian MF ANI calls.
 - Type **n** in the **ANI Req** field if you do *not* require ANI on incoming R2-MFC or Russian MF ANI calls.
 - Type **r** if the **Allow ANI Restriction on AAR/ARS** field is set to **y** on the **Feature-Related System Parameters** screen. When the **ANI Req** field is set to **r**, the system drops a call on a Russian shuttle trunk or a Russian rotary trunk if the ANI request fails.
- 9 The **Percent Full** field displays the percent of the allocated system memory that is used by AAR/ARS. You cannot change this field.
- 10 Press **Enter** to save your changes.

Administering the AAR Digit Analysis table

To administer the AAR Digit Analysis Table for your system:

- 1 Type **change aar analysis n**, where **n** is the number of the AAR Analysis table. Press **Enter**.
The system displays the **AAR Digit Analysis Table** screen ([Figure 299, AAR Digit Analysis Table screen](#), on page 1130).

Figure 299: AAR Digit Analysis Table screen

AAR DIGIT ANALYSIS TABLE						
						Percent Full: 2
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd
111	7	7	111	aar		n
2	7	7	999	aar		n
200	7	7	200	aar		n
201	7	7	201	aar		n
202	7	7	202	aar	13	n
203	7	7	203	aar		n
204	7	7	204	aar		n
205	7	7	205	aar		n
220	6	8	220	aar		n
222	7	7	222	aar		n
230	7	7	30	aar		n
241	7	7	3	aar		n
242	7	7	42	aar		n
244	7	7	4	aar		n
245	7	7	45	aar		n

- 2 In the **Dialed String** field, type the numbers that the system compares to the number that the user dials.

The system uses the dialed string that most closely matches the number that the user dials. For example, if a user dials 297-1234, and the AAR Digit Analysis Table contains a dialed string of 297-1 and a dialed string of 297-123, the system uses the dialed string 297-123 as the match.

When the system compares a number dialed by the user with a dialed string entry of the same number and a dialed string entry with a wildcard character some of the numbers, the system chooses the exact match. For example, if a user dials 424, and the AAR Digit Analysis Table contains a dialed string of 424 and a dialed string of X24, the system uses the dialed string 424 as the match.

You can enter digits **0** through **9** and wildcard characters ***** or **x** or **X**. You can enter a maximum of 18 digits.

- 3 In the **Min** field, type the minimum number of user-dialed digits that the software uses to compare with the dialed string in the **Dialed String** field. You can type **1** to the number in the **Max** field.
- 4 In the **Max** field, type the maximum number of user-dialed digits the software uses to compare with the dialed string in the **Dialed String** field. You can type from the number in the **Min** field to **28**.
- 5 In the **Route Pattern** field, type the route information that you want the server to use for the dialed string in the **Dialed String** field. Perform one of the following actions:
 - Type **p1** to **p2000** to specify the route index number established on the **Partition Routing Table** screen
 - Type **1** to **640** to specify the route pattern uses to route the call
 - Type **1** to **999** to specify the route pattern used to route the call. Use this number range only for the S8300 Media Server.
 - Type **r1** to **r32** to specify the remote home numbering plan are table. You must type these entries if RHNPA translations are required for the corresponding dialed string.
 - Type **node** to designate node number routing.
 - Type **deny** to block the call.

- 6 In the **Call Type** field, type the call type associated with each dialed string. The call types indicate the numbering requirements on different trunk networks. Perform one of the following actions:
- Type **aar** for regular AAR calls.
 - Type **intl** for if the Route Index contains public network ISDN trunks that require the international type of number encodings.
 - Type **pubu** if the Route Index contains public network ISDN trunks that require an unknown type of number encodings.
 - Type **lev0**, **lev1**, or **lev2** to specify the ISDN Private Number Plan (PNP) number formats.
- For more information on ISDN Numbering—Private, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

The ISDN protocols are as follows:

Table 107: ISDN protocols

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)
lev3	PNP(9)	Regional Level 2(1)

- 7 In the **Node Number** field, type the number of the destination node in a private network if you use node number routing or Distributed Communication System (DCS).
- 8 Use the **ANI Req** field only if the **Request Incoming ANI (non-AAR/ARS)** field on the **Mutifrequency-Signaling-Related System Parameters** screen is set to **n**. If the **Request Incoming ANI (non-AAR/ARS)** field on the **Mutifrequency-Signaling-Related System Parameters** screen is set to **n**, perform one of the following actions:
- Type **y** in the **ANI Req** field if you require ANI on incoming R2-MFC or Russian MF ANI calls.
 - Type **n** in the **ANI Req** field if you do *not* require ANI on incoming R2-MFC or Russian MF ANI calls.
 - Type **r** if the **Allow ANI Restriction on AAR/ARS** field is set to **y** on the **Feature-Related System Parameters** screen. When the **ANI Req** field is set to **r**, the system drops a call on a Russian shuttle trunk or a Russian rotary trunk if the ANI request fails. Trunks other than Russian shuttle trunks and Russian rotary trunks process an **r** entry in the **ANI Req** field as a **y**.
- 9 The **Percent Full** field displays the percent of the allocated system memory that is used by AAR/ARS. You cannot change this field.
- 10 Press **Enter** to save your changes.

Administering the ARS Digit Analysis table

To administer the ARS Digit Analysis Table for your system:

- 1 Type **change ars analysis *n***, where *n* is the number of the ARS Analysis table. Press **Enter**.

The system displays the **ARS Digit Analysis Table** screen ([Figure 300, ARS Digit Analysis Table screen](#), on page 1132).

Figure 300: ARS Digit Analysis Table screen

ARS DIGIT ANALYSIS TABLE						
Location: all				Percent Full: 2		
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd
101xxxx0	8	8	deny	op		n
101xxxx0	18	18	deny	op		n
101xxxx01	16	24	deny	iop		n
101xxxx011	17	25	deny	intl		n
101xxxx1	18	18	deny	fnpa		n
10xxx0	6	6	deny	op		n
10xxx0	16	16	deny	op		n
10xxx01	14	22	deny	iop		n
10xxx011	15	23	deny	intl		n
10xxx1	16	16	deny	fnpa		n
11	2	2	10	svcl		n
120	11	11	deny	fnpa		n
1200	11	11	deny	fnpa		n
121	11	11	deny	fnpa		n
122	11	11	deny	fnpa		n

- 2 The **location** field is a display-only field.

If the **ARS** field and the **Multiple Locations** fields on the **Optional Features** screen are set to **y**, the **location** field is set to the numbers **1** through **64**. The numbers 1 through 64 in the **location** field define the location of the server that uses this ARS Digit Analysis Table.

If the **Multiple Locations** field on the **Optional Features** screen is set to **n**, the **location** field is set to **all**. A value of **all** in the **location** field indicates that this ARS Digit Analysis Table is the default for all port network locations.

- 3 In the **Dialed String** field, type the numbers that the system compares to the number that the user dials.

The system uses the dialed string that most closely matches the number that the user dials. For example, if a user dials 297-1234, and the ARS Digit Analysis Table contains a dialed string of 297-1 and a dialed string of 297-123, the system uses the dialed string 297-123 as the match.

When the system compares a number dialed by the user with a dialed string entry of the same number and a dialed string entry with a wildcard character some of the numbers, the system chooses the exact match. For example, if a user dials 424, and the ARS Digit Analysis Table contains a dialed string of 424 and a dialed string of X24, the system uses the dialed string 424 as the match.

You can enter digits **0** through **9** and wildcard characters ***** or **x** or **X**. You can enter a maximum of 18 digits.

- 4 In the **Min** field, type the minimum number of user-dialed digits that the software uses to compare with the dialed string in the **Dialed String** field. You can type **1** to the number in the **Max** field.
- 5 In the **Max** field, type the maximum number of user-dialed digits the software uses to compare with the dialed string in the **Dialed String** field. You can type from the number in the **Min** field to **28**.
- 6 In the **Route Pattern** field, type the route information that you want the server to use for the dialed string in the **Dialed String** field. Perform one of the following actions:
 - Type **p1** to **p2000** to specify the route index number established on the **Partition Routing Table** screen
 - Type **1** to **640** to specify the route pattern uses to route the call
 - Type **1** to **999** to specify the route pattern used to route the call. Use this number range only for the S8300 Media Server.
 - Type **r1** to **r32** to specify the remote home numbering plan are table. You must type these entries if RHNPA translations are required for the corresponding dialed string.
 - Type **node** to designate node number routing.
 - Type **deny** to block the call.
- 7 In the **Call Type** field, type the call type associated with each dialed string. The call types indicate the numbering requirements on different trunk networks. Perform one of the following actions:
 - Type **alrt** to alert attendant consoles or other digital telephone when a user places and emergency call.
This a normal China number 1 call type.
 - Type **intl emer** to designate an emergency call.
This a normal China number 1 call type.
 - Type **fnpa** to designate a 10-digit North American Numbering Plan (NANP) call (11 digits with a Prefix Digit of 1).
This an attendant China number 1 call type.
 - Type **hnpa** to designate a 7-digit NANP call.
This a normal China number 1 call type.
 - Type **intl** to designate a public-network international number.
This a toll-auto China number 1 call type.
 - Type **lop** to designate an international operator.
This an attendant China number 1 call type.
 - Type **natl** to designate non-NANP.
This a normal China number 1 call type.
 - Type **npvt** to designate national private.
This a normal China number 1 call type.
 - Type **nsvc** to designate national service.
This a normal China number 1 call type.
 - Type **op** to designate an operator.
This an attendant China number 1 call type.

- Type **pubu** to designate public-network number (E.164)-unknown.
This a normal China number 1 call type.
 - Type **svcl** to designate national(2).
This a toll-auto China number 1 call type.
 - Type **svct** to designate national(2).
This a normal China number 1 call type.
 - Type **svft** to designate a service call, first part control.
This a local China number 1 call type.
 - Type **svfl** to designate a service call, first part control.
This a toll China number 1 call type.
- 8** In the **Node Number** field, type the number of the destination node in a private network if you use node number routing or Distributed Communication System (DCS).
- 9** Use the **ANI Req** field only if the **Request Incoming ANI (non-AAR/ARS)** field on the **Mutifrequency-Signaling-Related System Parameters** screen is set to **n**. If the **Request Incoming ANI (non-AAR/ARS)** field on the **Mutifrequency-Signaling-Related System Parameters** screen is set to **n**, perform one of the following actions:
- Type **y** in the **ANI Req** field if you require ANI on incoming R2-MFC or Russian MF ANI calls.
 - Type **n** in the **ANI Req** field if you do *not* require ANI on incoming R2-MFC or Russian MF ANI calls.
 - Type **r** if the **Allow ANI Restriction on AAR/ARS** field is set to **y** on the **Feature-Related System Parameters** screen. When the **ANI Req** field is set to **r**, the system drops a call on a Russian shuttle trunk or a Russian rotary trunk if the ANI request fails. Trunks other than Russian shuttle trunks and Russian rotary trunks process an **r** entry in the **ANI Req** field as a **y**.
- 10** The **Percent Full** field displays the percent of the allocated system memory that is used by AAR/ARS. You cannot change this field.
- 11** Press **Enter** to save your changes.

Administering the extension number portability numbering plan

To administer the extension number portability numbering plan for your system:

- 1** Type **change enp-number-plan**. Press **Enter**.

The system displays the **Extension Number Portability Numbering Plan** screen ([Figure 301, Extension Number Portability Numbering Plan screen](#), on page 1135).

Figure 301: Extension Number Portability Numbering Plan screen

EXTENSION NUMBER PORTABILITY NUMBERING PLAN									
ENP Code	ENP Code	ENP Code	ENP Code	ENP Code	ENP Code	ENP Code	ENP Code	ENP Code	ENP Code
0x:	1x:	2x:	3x:	4x:	5x:	6x:	7x:	8x:	9x:
00:	10:	20:	30:	40:	50:	60:	70:	80:	90:
01:	11:	21:	31:	41:	51:	61:	71:	81:	91:
02:	12:	22:	32:	42:	52:	62:	72:	82:	92:
03:	13:	23:	33:	43:	53:	63:	73:	83:	93:
04:	14:	24:	34:	44:	54:	64:	74:	84:	94:
05:	15:	25:	35:	45:	55:	65:	75:	85:	95:
06:	16:	26:	36:	46:	56:	66:	76:	86:	96:
07:	17:	27:	37:	47:	57:	67:	77:	87:	97:
08:	18:	28:	38:	48:	58:	68:	78:	88:	98:
09:	19:	29:	39:	49:	59:	69:	79:	89:	99:

For more information on the *Extension Number Portability Numbering Plan* screen, [click here](#), or see the *Administrator's Guide for Network Connectivity for Avaya MultiVantage Software*.

Reports for Uniform Dial Plan

The following reports provide information about the Uniform Dial Plan capability:

- The Uniform Dial Plan report shows the details of the Dial Plan.

Considerations for Uniform Dial Plan

This section provides information about how the Uniform Dial Plan (UDP) capability behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Uniform Dial Plan under all conditions.

- In North American network environments, the system routes extensions that start with 0 to an attendant. Avaya recommends that you use a digit other than 0, as the leading digit when you assign extension numbers.
- When you call an extension number on another server, you might experience slight delay before you hear call-progress tones. This delay is caused by the trunk signaling that is necessary to complete the call to the remote media server or to the switch.
- When you select the option that causes the software to look at the UDP table first, the system routes calls, that might otherwise terminate at a local extension, over the network. If you do not want the system to route calls over the network, you must remove the extensions from the UDP table. Once you remove the extensions from the UDP table, users can dial the local extension.

- If Automatic Alternate Routing (AAR) is active, the software sends facility restriction levels (FRL) and Traveling Class Marks (TCM) with the private network number. UDP Code and AAR Code conversions use the FRL assigned to the caller. Extension number portability (ENP) Node conversion always raises the FRL to the maximum of seven.

If an FRL is insufficient to access the facility, the software denies access. The software does not prompt for an authorization code, even if authorization codes are enabled and administered in your system.

- If AAR is not active, do not equip your system to use tandem-tie trunks to transport UDP numbers. You should not use tandem-tie trunks to transport UDP numbers, because the termination server does not recognize the TCM.

Avaya recommends that you never use tandem-tie trunks to transport UDP numbers. When you use tandem-tie trunks to transport UDP numbers, the receiving media server or switch does not recognize the TCM and the hop count that follow the extension.

Interactions for Uniform Dial Plan

This section provides information about how the Uniform Dial Plan (UDP) capability interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Uniform Dial Plan in any capability configuration.

- Automatic Alternate Routing (AAR)
AAR routes UDP calls. The AAR subset is included with the UDP. If AAR and UDP are both enabled on your system, the 7-digit AAR number provides the same routing as the UDP.
- Dial Plan
 - Extension numbers on a server do not need to be part of a UDP. When extension numbers are not part of a UDP, the server software uses a non-UDP to handle calls that are associated with those extensions.
 - When you administer the Dial Plan and designate a group of extensions as UDP nonlocal, you can specify either that the software search for local extensions first, or search for local extensions last.
- DID trunk group
DID calls to a UDP extension number might require the DID trunk group to insert the necessary digits to create the full extension number. For example, if the DID trunk provides 4 digits, but a 5-digit UDP is in place, the DID trunk group must insert the appropriate leading digit.
- Distributed Communications System (DCS)
UDP is required when DCS is provided. The necessary UDP software is provided with the DCS software.
- Extension Number Portability (ENP)
If you administer a user extension to use ENP node routing, ENP routes the call to the correct server.
If you enable both AAR and UDP, the 7-digit AAR number provides the same routing as UDP. The 7-digit AAR number uses ENP to route the call.

Visually Impaired Attendant Service

Use the Visually Impaired Attendant Service (VIAS) feature to provide voice feedback to a visually impaired attendant.

Detailed description of Visually Impaired Attendant Service

This section provides a detailed description of the Visually Impaired Attendant Service (VIAS) feature.

You can use Visually Impaired Attendant Service (VIAS) to provide voice feedback to a visually impaired attendant. Each voice phrase is a sequence of one or more single-voiced messages. VIAS defines six buttons for the visually impaired attendant:

- Visually impaired service activation and deactivation
This button activates or deactivates VIAS. When you press the button, the system reenables all ringers that are disabled, for example, recall and incoming call ringers.
- Console status
When you press this button, the system voices the status of the:
 - Console, for example, busy, available, or night service
 - Attendant queue
 - System alarms
- Display status
When you press this button, the system voices the information that is on the console display. VIAS is unavailable for some displays, such as Class of Restriction (COR), restriction information, personal names, and the purpose of some calls.
- Last operation
When you press this button, the system voices the last operation that was performed at the console.
- Last voice message
When you press this button, the system voices the most recent message at the console.
- Direct trunk group selection status
When you press this button, the system voices the status of an attendant-monitored trunk group.

A visually impaired attendant can use the Inspect mode to locate each button, to determine the feature that is assigned to the button.

Hardware requirements for Visually Impaired Attendant Service

The Visually Impaired Attendant Service (VIAS) feature requires the following hardware:

- An attendant console
- A speech synthesizer circuit pack

Administering Visually Impaired Attendant Service

This section contains prerequisites and the screens for administering the Visually Impaired Attendant Service (VIAS) feature.

Prerequisites for administering Visually Impaired Attendant Service

You must complete the following actions before you can administer the Visually Impaired Attendant Service (VIAS) feature:

- Set up an attendant console. For information on how to set up an attendant console, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Screens for administering Visually Impaired Attendant Service

Screen name	Purpose	Fields
<i>Attendant Console</i>	Administer the Visually Impaired Attendant Service (VIAS) buttons: • vis • con-stat • display • dtgs-stat • last-mess • last-op	Any available button field in the FEATURE BUTTON ASSIGNMENTS area

Reports for Visually Impaired Attendant Service

The following reports provide information about the Visually Impaired Attendant Service (VIAS) feature:

- None

Considerations for Visually Impaired Attendant Service

This section provides information about how the Visually Impaired Attendant Service (VIAS) feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Visually Impaired Attendant Service (VIAS) under all conditions. The following considerations apply to Visually Impaired Attendant Service (VIAS):

- None

Interactions for Visually Impaired Attendant Service

This section provides information about how the Visually Impaired Attendant Service (VIAS) feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Visually Impaired Attendant Service (VIAS) in any feature configuration.

- Audible Message Waiting

The system generates a stutter tone before the dial tone, when a message waits at an extension. A visually impaired attendant can use the stutter tone, instead of a message light, to detect a message that waits at an extension.

- Auto Start and Don't Split

If you activate or deactivate VIAS while Auto Start and Don't Split is active, the system deactivates Auto Start and Don't Split.

- Automatic Circuit Assurance (ACA)

When the attendant presses the display button, the system voices:

- “Automatic Circuit Assurance,” if an ACA call is unanswered
- “Automatic Circuit Assurance,” and the extension that is assigned to the ACA call, if the call is answered

- Malicious Call Trace (MCT)

The system voices displays of MCT activation, but not of MCT control.

Voice Message Retrieval

Use the Voice Message Retrieval feature to allow attendants, telephone users, and remote access users to retrieve Leave Word Calling (LWC) messages and Call Coverage messages.

Detailed description of Voice Message Retrieval

This section provides a detailed description of the Voice Message Retrieval feature.

Voice Message Retrieval is only used for the retrieval of messages. When a telephone is in Voice Message Retrieval mode, you cannot use the telephone to make calls or access other features. You can use Voice Message Retrieval to retrieve your own messages, or messages for another user. However, a different user's messages for another user can only be retrieved at a telephone or an attendant console that is in the coverage path. This is done by an administered system-wide message retriever, or by a remote-access user when the extension and the associated security code are known. The number of simultaneous Voice Message Retrieval users that are possible depends on the number of speech-synthesizer circuit packs in the system.

You can designate certain telephones and attendants for system-wide message retrieval. These telephones are the same as that you use for Display Message Retrieval, and have the same privileges. Voice Message Retrieval cannot be accessed from rotary phones.

You can restrict unauthorized users from retrieving messages. Use the Lock function to restrict a telephone and the Unlock function to release the restriction. To activate Lock, users dial a system-wide access code. To cancel Lock, users dial a system-wide access code, and then an Unlock security code that is unique to the telephone. These functions only apply to the telephone where the function is active. The system-wide access codes and security code that you use for the Lock and Unlock functions are the same as those use for LWC message retrieval by display. You can assign a status lamp to show the Lock status of the telephone.

Hardware requirements for Voice Message Retrieval

The Voice Message Retrieval feature requires the following hardware:

- None

Administering Voice Message Retrieval

This section describes the screens that you use to administer the Voice Message Retrieval feature.

Screens for administering Voice Message Retrieval

Screen name	Purpose	Fields
<i>Feature Access Code</i>	Set up Voice Message Retrieval.	• LWC Message Retrieval Lock • LWC Message Retrieval Unlock • Voice Coverage Message Retrieval Access Code • Voice Principal Message Retrieval Access Code
<i>Feature Related System Parameters</i>	Administer Voice Message Retrieval.	• Stations With System-Wide Retrieval Permission Message • Waiting Lamp Indicates Status
<i>Station</i>	Set up Voice Message Retrieval.	Security Code

Reports for Voice Message Retrieval

The following reports provide information about the Voice Message Retrieval feature:

- None

Considerations for Voice Message Retrieval

This section provides information about how the Voice Message Retrieval feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Voice Message Retrieval under all conditions. The following considerations apply to Voice Message Retrieval:

- None

Interactions for Voice Message Retrieval

This section provides information about how the Voice Message Retrieval feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Voice Message Retrieval in any feature configuration.

- AUDIX Interface

Retrieval of Leave Word Calling (LWC) messages by way of Voice Message Retrieval is separate and distinct from retrieval of messages by way of INTUITY AUDIX. LWC messages on INTUITY AUDIX cannot be accessed using Voice Message Retrieval. However, the user who calls Voice Message Retrieval is informed of any new messages for the principal on AUDIX:

- The Voice Message Retrieval voices that there are INTUITY 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: LWC messages with Voice Message Retrieval and all other messages with INTUITY AUDIX.

If you do not have a TN725B speech-synthesizer board, non-display telephone users cannot retrieve LWC messages sent by way of the LWC button on a telephone.

- Bridged Call Appearance

Voice Message Retrieval on a Bridged Call Appearance functions the same as if the feature were activated by the primary extension that is associated with the bridged call appearance.

- Leave Word Calling (LWC)

Voice Message Retrieval enhances LWC by allowing any authorized touchtone telephone user to retrieve messages.

Whisper Paging

Use the Whisper Paging feature to allow one user to interrupt or “bargue in” on the call of another user and make an announcement. The paging user dials a feature access code (FAC) or presses a feature button, and then dials the extension of the other user.

Detailed description of Whisper Paging

This section provides a detailed description of the Whisper Paging feature.

With the Whisper Paging feature, the page is unique because only the person on the paged extension can hear the page. Other parties on the call cannot hear the page, and the person making the page cannot hear anyone on the call. If the paged user has a display telephone, the paged user can see who makes the whisper page.

For example, users A and B are on a call. User C has an urgent message for user A and makes a whisper page. All three users hear the tone that signals the page, but only user A hears the page itself. User c who makes cannot hear users A or B.

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, if Call Forwarding- All Calls is activated on a telephone on which no active calls exist, a whisper page to that telephone rings as a priority call.

Group answering environments

Whisper Paging does not work with extensions that are assigned to a group answering environment. You cannot place a whisper page to the main extension assigned to a hunt group, a split, skill, or terminating extension group (TEG). 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 (DCS) networks or electronic tandem networks. Both the paging user and paged user must be on the same server that runs Avaya Communication Manager.

Speakerphones

When a call is on the speaker, an incoming whisper page is also heard over the speaker too. When the group listening feature on a 6400-series telephone is active, an incoming whisper page is heard on both the handset and the speaker.

Hardware requirements for Whisper Paging

The Whisper Paging feature requires the following hardware:

- None

Administering Whisper Paging

The following steps are part of the administration process for the Whisper Paging feature:

This section describes:

- Any prerequisites for administering the Whisper Paging feature
- The screens that you use to administer the Whisper Paging feature

Prerequisites for administering Whisper Paging

You must complete the following actions before you can administer the Whisper Paging feature:

- The server that runs Avaya Communication Manager must have a circuit pack that supports whisper paging. See the *Hardware Guide for Avaya™ Communication Manager* for information on specific models.
- Users must have 6400-, 7400-, 8400-, or 9400-series DCP (digital) phones.

Screens for administering Whisper Paging

Screen name	Purpose	Fields
<i>Station</i>	Activate Whisper Paging buttons.	- Whisper Page Activation - Answerback - Whisper Page Off
Feature Access Code	Activate a feature access Code (FAC) to Whisper Page.	Whisper Page Activation Access Code

End-user procedures for Whisper Paging

End users must perform specific procedures to use certain features. End users can activate or deactivate certain system features and capabilities. End users can also modify or customize some aspects of the administration of certain features and capabilities. This section includes the following end-user procedures for Whisper Paging:

Making 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.

The Do Not Disturb and Privacy - Attendant Lockout features can also block incoming whisper pages.

Reports for Whisper Paging

The following reports provide information about the Whisper Paging feature:

- None

Considerations for Whisper Paging

This section provides information about how the Whisper Paging feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Whisper Paging under all conditions. The following considerations apply to Whisper Paging:

- None

Interactions for Whisper Paging

This section provides information about how the Whisper Paging feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Whisper Paging in any feature configuration.

- 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 start 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 other 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 cannot 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 cannot 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 does not work. The opposite is true as well. If you activate Go to Cover and then press the whisper page activation button, you will not be 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 paging party and paged party must be on the same media server 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:

System administrators must ensure that this feature is managed appropriately in systems with tenant partitioning. Some tenants might not want other tenants to interrupt their calls.

- Transfer

A call cannot be transferred during an active whisper page.

- Vector Directory Number (VDN)

You cannot make a whisper page to a VDN. Pages must be made to a physical extension.

Wideband Switching

Use the Wideband Switching feature to dedicate two or more ISDN-PRI B-channels or DS0 endpoints for applications that require large bandwidth. Wideband Switching 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.

Detailed description of Wideband Switching

This section provides a detailed description of the Wideband Switching feature.

ISDN-PRI and the emulation of ISDN-PRI by Asynchronous Transfer Mode-Circuit Emulation Service (ATM-CES) divides a T1 or an 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. 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 Communication Manager serves as a gateway to many types of high-bandwidth traffic. In addition, DS1 converters are also used for wideband switching at remote locations.

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
- Wide Area Network (WAN) disaster recovery
- Scheduled batch processing (for example, nightly file transfers)
- Local Area Network (LAN) interconnections and imaging
- Other high bandwidth applications that involve high-speed data transmission, video transmission, and so on.

Channel allocation

The following table provides information on Wideband Switching channel types.

Channel type	Number of channels	Data rate (kbps)
H0	6	384
H11	24	1536
H12	30	1920
NXDS0 (T1)	2-24	128–1536
NXDS0 (E1)	2-31	128–1984

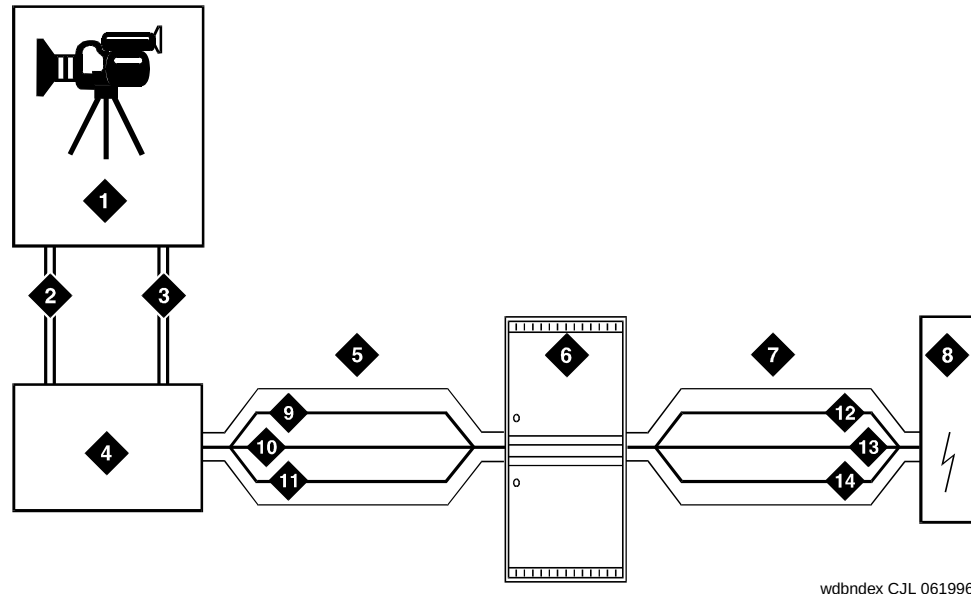
Perform wideband line-side channel allocation with one of three allocation algorithms:

- Fixed allocation - Provides contiguous-channel aggregation. The starting channel is constrained to a predetermined starting point. Used for only 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. [Figure 302, Typical video broadband application](#), on page 1153 shows an example.

Figure 302: Typical video broadband application



wdbndex CJL 061996

Figure notes

1	Video application	8	Network
2	Port 1	9	DS0 24 D-channel
3	Port 2	10	DS0 23 unused
4	ISDN terminal adaptor	11	DS0 1–6 wideband
5	Line-side ISDN-PRI	12	DS0 24 D-channel
6	Avaya Media Server	13	DS0 7–23 narrow bands
7	ISDN or ATM-CES trunk	14	DS0 1–6 wideband

Endpoint applications with signaling

An endpoint application is the origination point or the destination point of a wideband call. Endpoint applications can be any number of data applications, based on the needs of the customer.

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 Communication Manager 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. These messages indicate the data rate and the specific B-channels (DS0) to be used. The terminal adapter communicates all other call status information by way of standard ISDN messages. See *DEFINITY Line-Side ISDN Primary Rate Interface Technical Reference* for more information.

Line-side (T1 or E1) ISDN-PRI facilities

A line-side ISDN-PRI (T1 or E1) facility is comprised of a group of DS0s. 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 server running Communication Manager 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 to which an extension is assigned.

A PRI-endpoint can support calls of lower bandwidth. In other words, a PE having a width six DS0 channels can handle a call of one channel (64 Kbps) up to and including six channels (384 Kbps). Also, a PE can support calls on nonadjacent channels. For example, an endpoint application that is 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 the 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.

Nonsignaling endpoint applications

Wideband Switching can also support configurations that use nonsignaling (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 passes the call to the endpoint application. Unlike terminal adapters, the device service unit (DSU)/channel service unit (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 server. 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 media server. These endpoints simply transmit and receive wideband data when the connection is active.

NOTE:

Communication Manager 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 Switching and the components of Wideband Switching 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

The following table lists some key customer data rates.

Channel type	Number of channels	Data rate (Kbps)
H0	6	384
H11	24	1536

Channel type	Number of channels	Data rate (Kbps)
H12	30	1920
NXDS0 (T1)	2-24	128–1536
NXDS0 (E1)	2-31	128–1984

Data backup connection

Using Wideband Switching 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 switching 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 Communication Manager 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 server; ISDN endpoints can also establish additional calls when extra bandwidth is needed.

Any failures not automatically restored by Avaya Communication Manager are signaled to the endpoint application, which can initiate backup data connections over the same PRI endpoint. Avaya Communication Manager 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 software 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 a nonfacility-associated signaling (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 that comprise 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 Communication Manager receives and processes the SETUP message, it 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 Communication Manager 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.

Communication Manager 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
B-channel(s)	Wide	Outgoing call (to network) retried on another trunk	Outgoing call (to user) stays up
		No negotiation Incoming call (from network) dropped	No negotiation Incoming call (from user) dropped
	B-channel(s)	Outgoing calls (to network) stay up but likely are dropped by network because channels are in use, although there is a possibility some media servers or 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

1 of 2

GLARE RESOLUTION			
Outgoing Call Type	Incoming Call Type	Switch-Supporting User Protocol	Switch-Supporting Network Protocol
		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
		The incoming call from the network dropped	Incoming call (from user) dropped
		The outgoing calls to the 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 end because other end, if it is not Avaya Communication Manager, may let network call win.
			2 of 2

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 Communication Manager 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, Communication Manager 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

Hardware requirements for Wideband Switching

The Wideband Switching feature requires the following hardware:

- None

Administering Wideband Switching

The following steps are part of the administration process for the Wideband Switching feature:

This section describes:

- The screens that you use to administer the Wideband Switching feature

Screens for administering Wideband Switching

Screen name	Purpose	Fields
<i>Access Endpoint</i>	Administer Wideband Switching.	All
<i>System Parameters-Customer Options</i>	Administer Wideband Switching.	Wideband Switching
<i>PRI Endpoint</i>	Administer Wideband Switching.	All
<i>ISDN Trunk Group</i>	Administer Wideband Switching.	All
<i>Route Pattern</i>	Administer Wideband Switching.	All
<i>Fiber Link Administration</i>	Administer Wideband Switching.	All

Reports for Wideband Switching

The following reports provide information about the Wideband Switching feature:

- None

Considerations for Wideband Switching

This section provides information about how the Wideband Switching feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of Wideband Switching under all conditions. The following considerations apply to Wideband Switching:

- 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 for Wideband Switching

This section provides information about how the Wideband Switching feature interacts with other features on the system. Use this information to ensure that you receive the maximum benefits of Wideband Switching in any feature configuration.

- **Administered Connections**
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 (ACA)**
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 (CDR)**
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 (CMS) and Basic Call Management System (BCMS)**
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 (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 to a VDN, that sends the call to the first available PRI endpoint.

- **Class of Restriction (COR)**
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)**
COS determines the class of features that a wideband endpoint can activate.
- **Facility Associated Signaling (FAS) and Non-Facility Associated Signaling (NFAS)**
FAS and NFAS with or without D-Channel Backup requires administration by way of 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)**
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 Routing

Use the World Class Routing feature to direct outgoing calls. The World Class Routing feature has two capabilities: Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS).

- Automatic Alternate Routing (AAR) routes calls within your company over your own private network.

The system converts the number that the user dials. The system then analyzes the number, and routes the call as a private-network call.

- Automatic Route Selection (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.

The system converts the number that the user dials. The system then analyzes the number, and routes the call as a public-network call.

Automatic routing starts when a user dials a feature access code (FAC) and then the number that the user wants to call. The system analyzes the dialed digits, selects the route for the call, and deletes and inserts digits, if necessary. The system then routes the call over the trunks that you specify in your routing tables. AAR and ARS 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 AAR calls can be converted to ARS calls.

The FAC for AAR is usually the digit 8. The FAC for ARS is usually the digit 9 in the US and 0 outside the US. When your Avaya technician or business partner sets up AAR on your server that is running Avaya Communication Manager, he or she usually assigns the FAC for AAR. You can administer your own FAC for ARS. With Communication Manager 2.0, you can also use ARS without FAC.

Detailed description of World Class Routing

This section provides a detailed description of the World Class Routing feature:

- [Overview of automatic routing](#)
- [Understanding ARS analysis](#)
- [Examples of digit conversion](#)
- [ARS dialing without FAC](#)
- [AAR and ARS partitioning](#)

Overview of automatic routing

The figure shows an overview of automatic routing:

Figure 303: Automatic Routing Overview

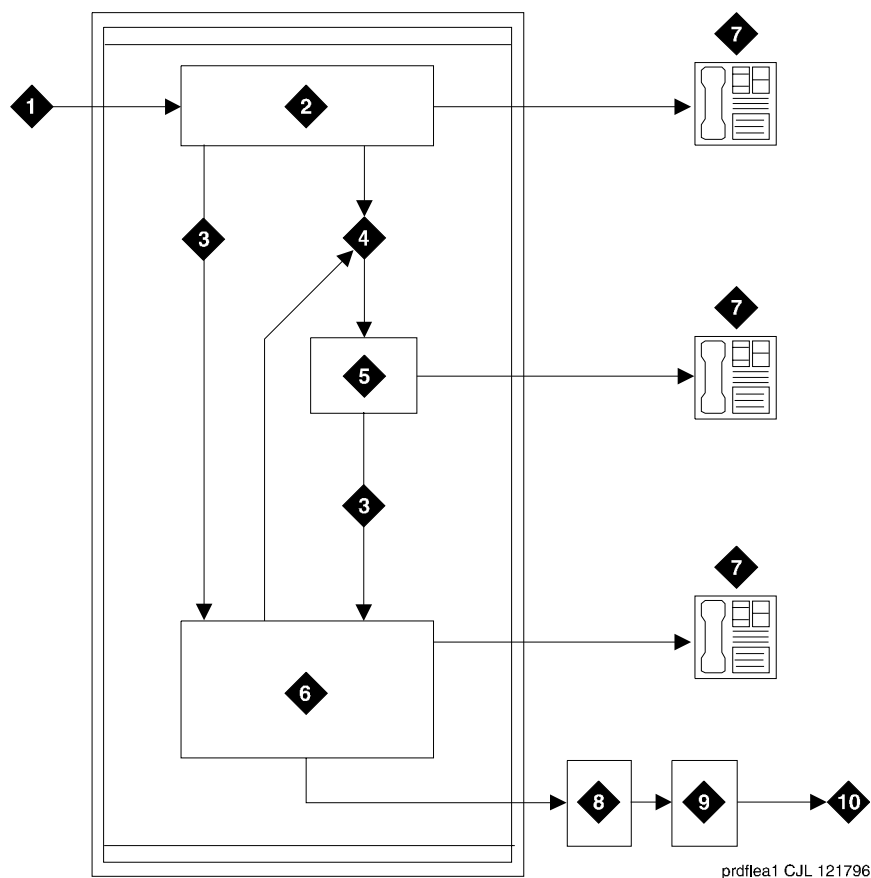


Figure notes

1	Receive input from a telephone, public network trunk, or private network trunk	7	Terminate the call at telephone
2	Analyze digits to determine the address type from Dial Plan Analysis Table	8	Analyze digits based on information from the AAR and ARS Digit Analysis Tables. Determine route pattern (Route Pattern, Node Number Routing, Extended Trunk Access screens)
3	Direct the call to AAR or ARS	9	Select an outgoing trunk group and delete and insert digits
4	Direct the call to the Uniform Dial Plan (UDP)	10	Send the call to a public network trunk or private network trunk
5	Use UDP to determine route		
6	Delete and insert digits based on AAR and ARS Digit Conversion Tables		

Understanding ARS analysis

With ARS, the system checks the digits in the called number against an ARS Digit Analysis Table to determine how to handle the dialed digits. The system also uses Class of Restriction (COR) and Facility Restriction Level (FRL) to determine the calling privileges.

[Figure 304, ARS Digit Analysis Table](#), on page 1167 shows a simple ARS digit analysis table.

Figure 304: ARS Digit Analysis Table

ARS DIGIT ANALYSIS TABLE							
		Location:		all	Percent Full: 6		
Dialed String	Total	Min	Max	Route Pattern	Call Type	Node Num	ANI Req
1	1	1	1	12	svcl	___	n
1	11	11	11	30	fnpa	___	n
1	12	23	17	intl	___	___	n
10xxx	5	5	5	deny	op	___	n
1800	11	11	11	30	fnpa	___	n
2	7	7	7	2	hnpa	___	n
3	7	7	7	2	hnpa	___	n
4	7	7	7	2	hnpa	___	n
5	7	7	7	2	hnpa	___	n
6	7	7	7	2	hnpa	___	n
7	7	7	7	2	hnpa	___	n

- Dialed string - Lists the first digits in the dialed string. When a user makes an outgoing call, the system analyzes the digits, and looks for a match in the table. The system then uses the information in the matching row to determine how to route the call.

For example, if a caller places a call to 1-303-233-1000, the system matches the dialed digits with the digits in the first column of the table. In this example, the dialed string matches the digit 1. Then the system matches the length of the entire dialed string (11 digits) to the minimum and the maximum (Max) length columns. In this example, the 11-digit call that started with 1 follows route pattern 30 as an fnpa call.

The first dialed digit for an external call is often an access code. If the digit 9 is defined as the ARS access code, the system drops this digit and uses the **ARS Analysis Table** to analyze the remaining digits.

- Route Pattern - Points to the route that handles the calls that match the dial string.
- Call Type - Describes what kind of call is made with this dial string. Call Type helps the system determine how to handle the dialed string.
- Node Num - States the number of the destination node in a private network if you are using node number routing or DCS. Valid values are 1 to 999 or blank.

For more information, [click here](#), or see the *Administrator's Guide for Avaya Communication Manager*.

Examples of digit conversion

The system uses the AAR or ARS Digit Conversion Table to change a dialed number for more efficient routing. Digits can be inserted or deleted from the dialed number. For instance, you can tell the system 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 ARS digit conversion examples in this section reflect the following values:

- ARS feature access code = 9
- AAR feature access code = 8
- Private Network Office Code (also known as Home RNX) = 222
- Also, in these examples, Prefix 1 is required on all long-distance DDD calls and Dashes (-) are for readability only.

The system maps the dialed digits to the pattern that most closely matches the dialed number. For example, if the dialed string is 957-1234 and the table contains the matching patterns 957-1 and 957-123, the match is on pattern 957-123.

ARS dialing without FAC

Table 108: 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 through AAR for RNX 362
Long distance call to specified carrier	9-10222+DDD	10222	(blank)	(blank)	Call routes as dialed with DDD # over the private network
Terminating a local DDD call to an internal telephone	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 server (RNX 222), then to attendant (222-0111).
International call to announcement (This method can also be used to block unauthorized IDDD calls)	9-011-91-672530	011-91	222-1234#	222.1234-	Call routes to local server (RNX 222), then to announcement extension (222-1234).
International call from certain European countries that need dial tone detection	0-00-XXXXXXXX	00	+00+	00+XXX X	The first 0 indicates ARS, the second pair of 0s indicates an international call, the pluses denote "wait" for dial tone detection.

This feature allows users to place calls without the need to first dial a Feature Access Code (FAC), such as the number 9 to access an outside line. The system recognizes the call as an ARS or an AAR call, and accordingly uses the ARS or AAR digit analysis and digit conversion tables to route the call. This enhancement was added to the Communication Manger 2.0 release.

This feature provides for the following additional calling options:

- **Public-network dialing:** This feature eliminates the need to enter a FAC for access to a public network facility, or trunk. The typical application is apartments or assisted living complexes that want to provide a public telephone dialing look and feel to the end users.
- **UDP networks:** Many customers have large networks with 5-digit Uniform Dial Plans that have that have “run out of numbers.” This feature allows a customer to extend the network dial plan. A major disadvantage is that DCS feature transparency is only available with 4 and 5-digit dialing plans.
- **QSIG Networks:** The QSIG implementation does not require a 4-digit or a 5-digit Uniform Dial Plan like DCS. Providing an expanded private-network dial plan makes it easier to integrate the system into existing customer enterprise networks composed of non-Communication Manager systems. An expanded dial plan also facilitates those customers interested in converting a large DCS network to QSIG signaling protocol instead. DCS is not supported for private-network dial plans that are greater than 5-digits.
- **3-Digit UDP:** The ARS dialing without FAC allows 3-digit UDP. However, DCS does not work in a dial scheme of less than 4 digits. For feature transparency, QSIG must be used for signaling.

Avaya recommends that you use ARS or AAR Dialing without FAC only when dialing outside Central Office (CO) trunks. If you use this feature to expand dial plan lengths beyond 7-digits for internal extensions across a private network (QSIG or DCS) a variety of feature interactions might occur. Such interactions can include changes in information shown on display telephones, telephony features, and call center features. ARS/AAR Dialing without FAC has not been tested against all possible feature interactions in an intra-switch networked environment and therefore is not recommended for this use.

Extensions

A **dialable extension** is an extension that is can be dialed without going through ARS or AAR routing.

A **non-dialable extension** is an extension that can only be dialed using ARS or AAR routing. You cannot dial an non-dialable as administered. You must be dial it with a prefix as a longer number than the administered extension.

An **internal extension** is the normal extension (3-5 digits) as administered on a telephone.

Use the following general rules to determine how non-dialable extensions interact with a given feature:

- When an extension is dialed either directly from a telephone or during a feature invocation (e.g., abbreviated dialing, priority calling, call-forwarding, leave word calling), the ARS/AAR Dialing without FAC must be used.
- When a feature is being administered on behalf of a telephone user (for example, coverage paths, security violation notification), the internal extension must be entered through the administration tool.

AAR and ARS partitioning

You can use AAR and ARS partitioning to change the call routing plan for up to eight different user groups on a single server that is running Avaya Communication Manager. 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 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 trunk.

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 that have different routing because of special billing needs
- Groups that have dedicated use of a particular network facility
- Groups in different businesses that are 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 partitioned user groups.

Hardware requirements for World Class Routing

The World Class Routing feature requires the following hardware:

- None

Administering World Class Routing

The following steps are part of the administration process for the ARS feature of World Class Routing:

- [Managing calling privileges using COR and FRL](#)
- [Assigning the ARS FAC](#)
- [Setting up a location ARS FAC](#)
- [Displaying ARS analysis information](#)
- [Defining call types](#)
 - [Defining operator-assisted calls](#)
 - [Defining interexchange carrier calls](#)
- [Using restricted area codes and prefixes](#)
- [Using wildcards](#)

- [Defining local information calls](#)
- [Modifying call routing](#)
 - [Adding a new area code or prefix](#)
 - [Using ARS to restrict outgoing calls](#)
- [Defining ARS partitions](#)
 - [Setting up partition groups](#)
 - [Assigning a telephone to a partition group](#)
- [Setting up Time of Day Routing](#)

This section describes:

- Any prerequisites for administering the ARS feature
- The screens that you use to administer the ARS feature
- Complete administration procedures for the ARS feature

Prerequisites for administering World Class Routing

You must complete the following actions before you can administer the World Class Routing feature:

- None

Screens for World Class Routing

Screen name	Purpose	Fields
Station	Change the calling privileges of a telephone.	COR
Locations	Create location ARS FAC.	ARS Prefix 1 Required For 10-Digit NANP Calls?
AAR and ARS Digit Analysis Table	Define call types and defining inter-exchange carrier calls.	Dialed String
ARS Digit Analysis Table	Use wild cards.	• Dialed String • Total Min and Total Max • Route Pattern • Call Type

Screen name	Purpose	Fields
<i>ARS Digit Analysis Table</i>	Define local information calls.	• Dialed String • Total Min and Total Max • Route Pattern • Call Type
• <i>ARS Chosen Report</i> • <i>ARS Digit Analysis Table</i>	Add a new area code or a prefix.	• Total Min and Total Max • Route Pattern • Call Type • Node Num
<i>ARS Digit Analysis Table</i>	Restrict outgoing calls.	• Dialed String • Total Min and Total Max • Route Pattern • Call Type
• <i>ARS Chosen Report</i> • <i>Partition Routing Table</i>	Set up partition groups.	• Route Pattern • PGN
• <i>Class of Restriction Information</i> • <i>Authorization Code-COR Mapping</i>	Assign a phone to a partition group.	• COR description • Partition Group Number
<i>Time of Day Routing Plan</i>	Set up time of day routing.	• All • PGN

Managing calling privileges using COR and FRL

Each time that you set up a telephone, you use the **Station** screen to assign a Class of Restriction (COR). You can create different CORs for different groups of users. For example, you might 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. FRLs are ranked from 0–7, where 7 has the highest level of privileges.

You also assign an FRL to each route pattern preference on 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.

Suppose that you are setting up a telephone for a new executive. The current translations assign COR 1, with outward restrictions and an FRL 0, which is the lowest permission level available. You want to assign a COR with the highest level of permissions, FRL 7.

To change the COR of a telephone:

- 1 Type **change station *n***, where *n* is the telephone extension. Press **Enter**.

The system displays the **Station** screen ([Figure 305, Station screen](#), on page 1174).

Figure 305: Station screen

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: 3
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?	
Model:		
	Media Complex Ext:	
	IP Softphone? y	
	Remote Office Phone? n	

- 2 In the COR field, type the number of the new COR. In this example, type 7.
- 3 Press **Enter** to save your changes.

To change the FRL of this COR:

- 1 Type **change cor 7**. Press **Enter**.

The system displays the **Class of Restriction** screen ([Figure 306, Class of Restriction screen](#), on page 1175) for COR 7.

Figure 306: Class of Restriction screen

change cor 7
Page 1 of 4

CLASS OF RESTRICTION

COR Number: 7

COR Description: supervisor

FRL: 7 Can Be Service Observed? n Can Be A Service Observer? y Time of Day Chart: 1 Priority Queuing? n Restriction Override: none Restricted Call List? n Unrestricted Call List? _ _ _ _ _ Access to MCT? y Group II Category For MFC: 7 Send ANI for MFE? n_ MF ANI Prefix: _ Hear System Music on Hold? y	APLT? y Calling Party Restriction: none Called Party Restriction: none Forced Entry of Account Codes? n Direct Agent Calling? y Facility Access Trunk Test? n Can Change Coverage? n Fully Restricted Service? n Hear VDN of Origin Annc.? n Add/Remove Agent Skills? y Automatic Charge Display? n 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

2 In the FRL field, type the number of a new FRL. In this example, type 7.

3 Press **Enter** to save your changes.

All users with COR 7 now have the highest level of calling permissions.

Assigning the ARS FAC

The ARS Feature Access Code (FAC) must be set up on your system. In the U.S., the number 9 is usually the ARS FAC to make an outgoing call.

To assign the ARS FAC:

1 Type **change dialplan analysis**. Press **Enter**.

The system displays the *Dial Plan Analysis Table* screen ([Figure 307, Dial Plan Analysis Table screen](#), on page 1176).

Figure 307: Dial Plan Analysis Table screen

display dialplan analysis			DIAL PLAN ANALYSIS TABLE			Page 1 of 12		
						Percent Full: 7		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
00	2	attd	_____	-	_____	_____	-	_____
1	3	dac	_____	-	_____	_____	-	_____
2	4	ext	_____	-	_____	_____	-	_____
3	4	ext	_____	-	_____	_____	-	_____
3	1	aar	_____	-	_____	_____	-	_____
4	1	ars	_____	-	_____	_____	-	_____
4	5	ext	_____	-	_____	_____	-	_____
5	5	ext	_____	-	_____	_____	-	_____
5	7	ext	_____	-	_____	_____	-	_____
6	5	ext	_____	-	_____	_____	-	_____
7210	7	ext	_____	-	_____	_____	-	_____
8	7	ext	_____	-	_____	_____	-	_____
9	1	fac	_____	-	_____	_____	-	_____
*	3	fac	_____	-	_____	_____	-	_____
#	3	fac	_____	-	_____	_____	-	_____

- 2** In an empty row, type **9** for the dialed string, type **1** for the total length, and type **fac** for the call type.
- 3** Press **Enter** to save your changes.
- 4** Type **change features-access-codes**. Press **Enter**.

The system displays the *Feature Access Code (FAC)* screen ([Figure 308, Feature Access Code \(FAC\) screen](#), on page 1177).

Figure 308: Feature Access Code (FAC) screen

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:	_____
Attendant	Access Code:	_____
Auto Alternate Routing (AAR)	Access Code:	_____
Auto Route Selection (ARS)	Access Code 1:	9
	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:	_____

- 5 In the Auto Route Selection (ARS) Access Code 1 field, type **9**.
- 6 Press **Enter** to save your changes.

Setting up a location ARS FAC

Prerequisites

You must complete the following actions before you can set up a location ARS FAC:

- On the **Optional Features** screen, ensure that the **Multiple Locations** field is set to **y**. If this field is set to **y**, you can administer up to 250 locations depending on the configuration of the server running Communication Manager. If the **Multiple Locations** field is set to **n**, information for Location No. 1 applies to all of your locations.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

To set up a location ARS FAC:

- 1 Type **change locations**. Press **Enter**.

The system displays the **Locations** screen ([Figure 309, Locations screen](#), on page 1178).

Figure 309: Locations screen

change locations
Page 1 of x

LOCATIONS

ARS Prefix 1 Required For 10-Digit NANP Calls? y

Loc. No	Name	Timezone Offset	Rule	NPA	ARS FAC	Attd FAC	Pre-fix	Proxy Sel Rte pat
1	Main	00:00	0					
xxx		:						
xxx		:						
xxx		:						
xxx		:						
xxx		:						
xxx		:						
xxx		:						
xxx		:						
xxx		:						

- 2 In the ARS Prefix 1 Required For 10-Digit NANP Calls? field, type y.
- 3 Press **Enter** to save your changes.

NOTE:

The ARS access code on the **Feature Access Code (FAC)** screen is used when a location ARS does not exist. If a location ARS FAC exists, the ARS access code on the **Feature Access Code (FAC)** screen is denied from that location. By using a local ARS code, the ability to administer two ARS codes on the **Feature Access Code (FAC)** screen is lost.

Displaying ARS analysis information

The ARS Digit Analysis Table controls how the system routes calls. In this example, we look at how the system routes calls that begin with 1.

To display an ARS Digit Analysis Table:

- 1 Type **display ars analysis 1**. Press **Enter**.

The system displays the **ARS Digit Analysis Table** screen for dialed strings that begin with 1 ([Figure 310, ARS Digit Analysis Table screen](#), on page 1179).

Note that the system displays only as many dialed strings as can fit on one screen at a time.

Figure 310: ARS Digit Analysis Table screen

display ars analysis 1							Page 1 of X
ARS DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 3
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req	
101xxx0	8	8	deny	op	—	n	
101xxx0	18	18	deny	op	—	n	
101xxx01	16	24	deny	iop	—	n	
101xxx011	17	25	deny	intl	—	n	
101xxx1	18	18	deny	fnpa	—	n	
1010xxx	14	18	5	ixc	—	n	
10xxx0	6	6	deny	op	—	n	
10xxx0	16	16	deny	op	—	n	
10xxx01	14	22	deny	iop	—	n	
10xxx011	15	23	deny	intl	—	n	
10xxx1	16	16	deny	fnpa	—	n	
11	2	2	734	alrt	—	n	
120	11	11	deny	fnpa	—	n	
1200	11	11	deny	fnpa	—	n	
121	11	11	deny	fnpa	—	n	
122	11	11	deny	fnpa	—	n	

2 Press **Cancel** when you finish.

To see all the dialed strings that are defined for your system and run an ARS Digit Analysis report:

1 Type **list ars analysis**. Press **Enter**.

The system displays an **ARS Digit Analysis Table** screen for all dialed strings ([Figure 311, ARS Digit Analysis Table screen](#), on page 1179). You might want to print this report for your records.

Figure 311: ARS Digit Analysis Table screen

list ars analysis							Page 1 of X
ARS DIGIT ANALYSIS TABLE							
Location: all							
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num		
0	1	1	deny	op	—		
0	8	8	deny	op	—		
0	11	11	deny	op	—		
00	2	2	deny	op	—		
01	17	17	deny	op	—		
001	10	18	deny	op	—		
101xxx0	8	8	deny	op	—		
101xxx0	18	18	deny	op	—		
101xxx01	16	24	deny	iop	—		
101xxx011	17	25	deny	intl	—		
101xxx1	18	18	deny	fnpa	—		
10xxx0	6	6	deny	op	—		

2 Press **Cancel** when you finish.

Defining call types

To define call types, you must complete the following procedures:

- [Defining operator-assisted calls](#)
- [Defining interexchange carrier calls](#)

Defining operator-assisted calls

To define operator-assisted calls, the user first dials 9 to access ARS, then a 0, and the rest of the number.

To see how the system handles a call to an operator:

- 1 Type **display ars analysis 0**. Press **Enter**.

The system displays an **ARS Digit Analysis Table** screen for dialed strings that begin with 0 ([Figure 312, ARS Digit Analysis Table screen](#), on page 1180). The table in this example shows six translations for calls that begin with 0.

Figure 312: ARS Digit Analysis Table screen

display ars analysis 1							Page 1 of X	
ARS DIGIT ANALYSIS TABLE								
Location: all						Percent Full: 3		
	Dialed String	Total		Route	Call	Node	ANI	
		Min	Max	Pattern	Type	Num	Reqd	
0		1	1	deny	op	___	n	
0		8	8	deny	op	___	n	
0		11	11	deny	op	___	n	
00		2	2	deny	op	___	n	
01		17	17	deny	op	___	n	
001		10	18	deny	op	___	n	

- 2 Press **Cancel** when you finish.

Use the **ARS Digit Analysis Table** shown in [Figure 312, ARS Digit Analysis Table screen](#), on page 1180 and follow the routing for an operator-assisted call to New Jersey.

- 1 A user dials **9 0 908 956 1234**.
- 2 The system ignores the ARS FAC (**9** in this example), reviews the **ARS Digit Analysis Table** for **0**, and analyzes the number. Then the system:
 - Determines that the user dialed more than 1 digit
 - Determines that the user dialed 11 digits
 - Rules out the dialed strings for 00, 01, and 011
- 3 The system routes the call as an operator-assisted call.

Defining interexchange carrier calls

Interexchange carrier (IXC) numbers directly access your long-distance carrier lines. IXC numbers begin with 1010. After 1010, IXC numbers include 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 that the user first dials 9 to access ARS, then the rest of the number.

To route an ARS call to an IXC:

- 1 Type **display ars analysis 1**. Press **Enter**.

The system displays the *ARS Digit Analysis Table* screen appears for numbers that begin with 1 ([Figure 310, ARS Digit Analysis Table screen](#), on page 1179).

When you use **x** in the **Dialed String** field, the system recognizes **x** as a wildcard. The **x** represents any digit 0–9. If you dial 1010, the next 3 digits always match the **x** wildcard in the dialed string.

- 2 Press **Cancel** when you finish.

Use the *ARS Digit Analysis Table* shown in [Figure 310, ARS Digit Analysis Table screen](#), on page 1179 and follow the routing for an IXC call to AT&T. 1010288 is the carrier access code for AT&T.

- 1 A user dials **9 1010288**, plus a public network number.
- 2 The system ignores the ARS FAC (**9** in this example), reviews the *ARS Digit Analysis Table* for **1010**, and analyzes the number.
- 3 The system then matches **288** with **xxx**, and sends the call over route pattern 5.

Using restricted area codes and prefixes

Certain area code numbers are set aside in the North American Numbering Plan. For example, these numbers are 200, 300, 400, 500, 600, 700, 800, 877, 888, and 900. You must specifically deny calls made to area codes 200 through 900, except 800, 877, and 888.

If you do not want to incur charges, you can also deny access to the 976 prefix. The 976 prefix is set aside in each area code for pay-per-call services. You can block 976 or any other prefix in all Numbering Plan Areas (NPAs) with a single entry in the digit analysis table.

You can 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**. Press **Enter**.

The system displays the *AAR and ARS Digit Analysis Table* screen for calls that begin with 120 ([Figure 313, ARS Digit Analysis Table screen](#), on page 1182). The table in the example shows two translations for calls that begin with 120.

Figure 313: ARS Digit Analysis Table screen

change ars analysis 120							Page 1 of X	
ARS DIGIT ANALYSIS TABLE								
Location: all							Percent Full:	6
	Dialed	Total		Route	Call	Node	ANI	
	String	Min	Max	Pattern	Type	Num	Reqd	
120		11	11	4	fnpa	—	n	
1200		11	11	deny	fnpa	—	n	

- 2 Press **Cancel** when you finish.

Use the *ARS Digit Analysis Table* shown in [Figure 313, ARS Digit Analysis Table screen](#), on page 1182 for the next two examples.

Follow the routing for a long distance call that begins with 120, where the call is permitted. The 120 translation handles all dialed strings from 1-201 through 1-209.

- 1 A user dials **9 120**, plus 8 digits (the first of which is not **0**).
- 2 The system ignores the ARS FAC (**9** in this example), reviews the *ARS Digit Analysis Table* for **120**, and analyzes the number.
- 3 The system determines that the call is long distance and sends the call over Route Pattern 4.

Now follow the routing for a call that begins with the restricted area code 200:

- 1 A user dials **9 1200**, plus 7 digits.
- 2 The system ignores the ARS FAC (**9** in this example), reviews the *ARS Digit Analysis Table* for **1200**, and analyzes the number.
- 3 The system determines that the Call Type is **deny**, and the call does not go through.

Using wildcards

You can use wildcards to help control calls to certain numbers. When you use the wildcard **x** in the Dialed String field, the system recognizes **x** as any digit from 0 to 9. For example, you can use wildcards to restrict users from making calls to a 555 information operator where you might incur charges.

To prevent callers from placing calls to long distance 555 information numbers:

- 1 Type **change ars analysis 1**. Press **Enter**.
The system displays the *ARS Digit Analysis Table* screen for dialed strings that begin with 1 ([Figure 310, ARS Digit Analysis Table screen](#), on page 1179).
- 2 Move to a blank Dialed String field.
- 3 In the Dialed String field, type **1xxx555**.
- 4 In the Total Min and in the Total Max fields, type **11**.
- 5 In the Route Pattern field, type **deny**.
- 6 In the Call Type field, type **fnhp**.
- 7 Press **Enter** to save your changes.

Defining local information calls

To allow 411 service calls:

- 1 Type **change ars analysis 4** and press **Enter**.

The system displays the **ARS Digit Analysis Table** screen for dialed strings that begin with 4 ([Figure 314, ARS Digit Analysis Table screen](#), on page 1183).

Figure 314: ARS Digit Analysis Table screen

change ars analysis 4							Page 1 of X	
ARS DIGIT ANALYSIS TABLE								
Location: all							Percent Full:	6
	Dialed String	Total		Route	Call	Node	ANI	
		Min	Max	Pattern	Type	Num	Reqd	
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 Move to a blank Dialed String field.
- 3 In the Dialed String field, type **411**.
- 4 In the Total Min and in the Total Max fields, type **3**.
- 5 In the Route Pattern field, type **1**.
- 6 In the Call Type field, type **svcl** (service call).
- 7 Press **Enter** to save your changes.

Modifying call routing

To modify call routing, you must complete the following procedures:

- [Adding a new area code or prefix](#)
- [Using ARS to restrict outgoing calls](#)

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 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

To add a non local area code:

- 1 Type **list ars route-chosen**, the number **1**, the old area code, and the 7-digit number. For example, type **list ars route-chosen 14152223333**. Press **Enter**.

The system displays the **ARS Route Chosen Report** screen ([Figure 315, ARS Route Chosen Report screen](#), on page 1184).

Figure 315: ARS Route Chosen Report screen

list ars route-chosen 14152223333						Page 1 of X
ARS ROUTE CHOSEN REPORT						
Location: 1			Partitioned Group Number: 1			
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Number	Location
141	11	11	30	fnpa		all

- 2 Note the Total Min, Total Max, Route Pattern, and Call Type values on 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 Press **Cancel** when you finish.

Now add a new area code 650 that has the same values.

- 4 Type **change ars analysis 1650**. Press **Enter**.

The system displays the **ARS Digit Analysis Table** screen for dialed strings that begin with 1650 ([Figure 316, ARS Digit Analysis Table screen](#), on page 1184).

Figure 316: ARS Digit Analysis Table screen

change ars analysis 1650						Page 1 of X
ARS DIGIT ANALYSIS TABLE						
Location: all						Percent Full: 6
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd
1	11	11	30	fnpa	___	n
1650	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

- 5 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.
- 6 In the Dialed String field, type **1650**.
- 7 In the Total Min and Total Max fields, type the minimum and maximum values from step 2.
For this example, the minimum and maximum values are **11**.

- 8** In the **Route Pattern** field, type the route pattern from step 2.
For this example, the route pattern is **30**.
- 9** In the **Call Type** field, type the call type from step 2.
For this example, the route pattern is **fnpa**.
- 10** In the **Node Num** field, type the node number from step 2.
For this example, the node number is left blank.
- 11** Press **Enter** to save your changes.

To add a new prefix, follow the same directions to add a new area code. The one exception is to use a shorter dial string (such as **2223333**, where **222** is the old prefix) and a dial type of **hnpa**.

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** (instead of **11**) in the **Total Min** and the **Total Max** fields in step 6.

To see if the new area code or prefix number is set up as a toll call, type **display toll xxx**, where **xxx** is the prefix you want to review. Some users may not be allowed to dial toll call numbers.

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. In the U.S., you can restrict access to 900 and 976 pay-per-call numbers.

In the following example, Colombia is used as the country to restrict. The country code for Colombia is 57.

To prevent callers from placing calls to countries you want to restrict:

- 1** Type **change ars analysis 011xx**, where **011** is the international access, and **xx** is the country code of the restricted country. Press **Enter**.

The system displays the **ARS Digit Analysis Table** screen ([Figure 317, ARS Digit Analysis Table screen](#), on page 1186).

Figure 317: ARS Digit Analysis Table screen

change ars analysis 01157

Page 1 of X

ARS DIGIT ANALYSIS TABLE

Location: all

Percent Full: 6

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd
1	11	11	30	fnpa	___	n
1650	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
01157	10	23	deny	intl	___	n

- 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 In the Dialed String field, type **011xx**, where **xx** is the country code of the restricted country.
- 4 Type **10** in the Total Min field and type **23** in the Total Max field.
- 5 In the Route Pattern field, type **deny**.
- 6 In the Call Type field, type **intl**
- 7 Press **Enter** to save your changes.

Defining ARS partitions

You can use ARS partitioning to provide different call routing for a group of users, or for specific telephones.

If you used partitioning on a prior release of Avaya Communication Manager and you want to continue to use partitioning, please read this section carefully. In release 2.0 of Avaya Communication Manager, partition groups are defined on the **Partition Route Table**. Partition groups are no longer defined on the **Digit Analysis Table**.

Prerequisites

You must complete the following actions before you can define ARS partitions:

- On the **Optional Features** screen:
 - Ensure that the **Tenant Partitioning** field is set to **y**. If this field is not set to **y**, contact your Avaya representative for assistance.
 - Ensure that the **Time of Day Routing** field is set to **n**. If this field is not set to **n**, contact your Avaya representative for assistance.

To view the *Optional Features* screen, type **display system-parameters customer-options**. Press **Enter**.

Setting up partition groups

In this example, your company allows your employees to make local, long distance, and emergency calls. However, you have a telephone in the lobby 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, modify the routing for a partition group to enable only specific calls, such as U.S. based toll-free 1-800 calls. Then assign this partition group to the lobby phone.

To enable 1-800 calls for partition group 2:

- 1 Type **list ars route-chosen 18002221000**. Press **Enter**.
You can use any 7-digit number after the digits **1800** to create an example of the dialed string.
The system displays the *ARS Route Chosen Report* screen for partition group 1 ([Figure 318, ARS Route Chosen Report screen](#), on page 1187).

Figure 318: ARS Route Chosen Report screen

list ars route-chosen 18002221000						Page 1 of X
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 Note the route pattern for the selected dialed string. In this example, the route pattern for 1800 is **p1**. P1 indicates that the system uses the *Partition Routing Table* to determine what route pattern to use for each partition.
If there was a number with no “p” under *Route Pattern*, 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** when you finish.
- 4 Type **change partition-route-table index 1**. Press **Enter**.
The system displays the *Partition Routing Table* screens ([Figure 319, Partition Routing Table screen](#), on page 1188). In this example, partition group 1 can make 1800 calls, and these calls use route pattern 30.

Figure 319: Partition Routing Table screen

change partition-route-table index 1								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
-----	-----	-----	-----	-----	-----	-----	-----	-----
1	30	30	_____	_____	_____	_____	_____	_____
2	_____	_____	_____	_____	_____	_____	_____	_____
3	_____	_____	_____	_____	_____	_____	_____	_____
4	_____	_____	_____	_____	_____	_____	_____	_____
5	_____	_____	_____	_____	_____	_____	_____	_____
6	_____	_____	_____	_____	_____	_____	_____	_____
7	_____	_____	_____	_____	_____	_____	_____	_____
8	_____	_____	_____	_____	_____	_____	_____	_____
9	_____	_____	_____	_____	_____	_____	_____	_____
10	_____	_____	_____	_____	_____	_____	_____	_____
11	_____	_____	_____	_____	_____	_____	_____	_____
12	_____	_____	_____	_____	_____	_____	_____	_____
13	_____	_____	_____	_____	_____	_____	_____	_____
14	_____	_____	_____	_____	_____	_____	_____	_____
15	_____	_____	_____	_____	_____	_____	_____	_____

- 5 In the PGN 2 column that corresponds to Route Index 1, type **30**.

This entry tells the system to use route pattern 30 for partition group 2, and allow partition group 2 members to make calls to 1800 numbers.

- 6 Press **Enter** to save your changes.

Assigning a telephone to a partition group

Assigning a telephone extension to a partition group is a two step process. First, assign the partition group to a COR. Then assign that COR to the extension.

To assign a Class of Restriction (COR) to partition group 2:

- 1 Type **list cor**. Press **Enter**.

The system displays the *Class of Restriction Information* screen ([Figure 320, Class of Restriction Information screen](#), on page 1189).

Figure 320: Class of Restriction Information screen

list cor		CLASS OF RESTRICTION INFORMATION	Page 1 of 7
	COR	COR Description	
	0		
	1		
	2		
	3		
	4		
	5		
	6		
	7		
	8		
	9		
	10	Supervisor observer	
	11		
	12		
	13		
	14		

- 2 Choose a COR that has not been used.

In this example, we choose COR 3.

- 3 Press **Cancel** when you finish.

- 4 Type **change cor x**, where **x** is the COR number. Press **Enter**.

The system displays the **Class of Restriction** screen ([Figure 321, Class of Restriction screen](#), on page 1189).

Figure 321: Class of Restriction screen

change cor 3		CLASS OF RESTRICTION	Page 1 of 4
	COR Number: 3		
	COR Description: lobby		
	FRL: 0		APLT? y
	Can Be Service Observed? n		Calling Party Restriction: none
	Can Be A Service Observer? y		Called Party Restriction: none
	Partitioned Group Number: 2		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
	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

- 5 In the **COR Description** field, type a name for this COR.
For this example, the COR name is **lobby**.
- 6 In the **Partitioned Group Number** field, type the number of the partition group.
In this example, the partition group number is **2**.
- 7 Press **Enter** to save your changes.

To assign the COR to a telephone extension:

- 1 Type **change station n**, where **n** is the telephone extension. Press **Enter**.
The system displays the **Station** screen ([Figure 305, Station screen](#), on page 1174).
- 2 In the **COR** field, type the number of the COR.
In this example, the COR number is **3**.
- 3 Press **Enter** to save your changes.

Setting up Time of Day Routing

You can use Time of Day Routing to redirect calls to coverage paths according to the time of the day and the 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 the day of the week that 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.

Prerequisites

You must complete the following actions before you can set up Time of Day routing:

AAR or ARS must be administered on the system before you can use Time of Day Routing.

- On the **Optional Features** screen:
 - For AAR, ensure that either the **Private Networking** field or the **Uniform Dialing Plan** field is set to **y**. If neither field is set to **y**, contact your Avaya representative for assistance.
 - For ARS, ensure that both the **ARS** field is **y**, and the **Time of Day Routing** field is set to **y**. If both these fields are not set to **y**, contact your Avaya representative for assistance.

To view the **Optional Features** screen, type **display system-parameters customer-options**. Press **Enter**.

To display your Time of Day Routing Plan 1:

- 1 Type **display time-of-day x**, where **x** is the number of the routing plan. Press **Enter**.
The system displays the **Time of Day Routing Plan 1** screen ([Figure 322, Time of Day Routing Plan 1 screen](#), on page 1191).

Figure 322: Time of Day Routing Plan 1 screen

display time-of-day 1

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	__:	—	__:	—	__:	—	__:	—	__:	—

Note the routing plan that is currently in effect. In this example, this plan is for employees who can make only local calls.

You can see that in the example, two partition group numbers control Time-of-Day routing:

- PGN 1 begins 1 minute after midnight (00:01) every work day of the week until 8:00 a.m.
- PGN 2 begins at 8:00 a.m. every work day of the week until 12:00 p.m.
- PGN 1 begins at 12:00 p.m. every work day of the week until 1:00 p.m. (13:00).
- PGN 2 begins at 1:00 p.m. (13:00) every work day of the week until 5:00 p.m. (17:00).
- PGN 2 begins at 5:00 p.m. (17:00) every work day of the week until 12:00 a.m.
- PGN 1 is also used all day Saturday and Sunday.

2 Press **Cancel** when you finish.

To create a new Time-of-Day routing plan for long-distance calls for executives.

1 Type **change time-of-day 2**. Press **Enter**.

The system displays the *Time of Day Routing Plan 2* screen ([Figure 323, Time of Day Routing Plan screen](#), on page 1191).

Figure 323: Time of Day Routing Plan screen

change time-of-day 2										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	__:_	-	__:_	-	__:_	-	__:_	-	__:_	-			

2 Type **1** in each field as shown on *Time of Day Routing Plan 1*.

In this example, this is the PGN that is 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.

4 Press **Enter** to save your changes.

Now assign your new Time of Day Routing Plan 2 to the COR that is assigned to your executives.

In this example, the following conditions are true:

- 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.

Reports for World Class Routing

The following reports provide information about the World Class Routing feature:

- None

Considerations for World Class Routing

This section provides information about how the World Class Routing feature behaves in certain circumstances. Use this information to ensure that you receive the maximum benefits of the World Class Routing feature under all conditions:

- None

Interactions for World Class Routing

This section provides information about how the World Class Routing feature interact with other features on the system. Use this information to ensure that you receive the maximum benefits World Class Routing in any feature configuration:

- **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.
- **Call Forwarding**
The PGN of the forwarding party's COR is used to select the table to look up the route pattern.
- **Distributed Communications System (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.

Index

Symbols

#, [606](#), [665](#)
*, [606](#), [665](#), [920](#), [922](#)

Numerics

2420 DCP telephone, [233](#), [918](#)
4620 IP telephone, [233](#), [918](#)

A

AAR, *see* Automatic Alternate Routing (AAR)
Abbreviated Dialing (AD), [63](#)
 administering, [64](#)
 adding lists, [65](#)
 assigning telephones for group lists, [66](#)
 prerequisites, [64](#)
 commands
 add abbreviated-dialing group, [65](#)
 change station, [66](#)
 display system-parameters customer-options, [65](#)
 considerations, [68](#)
 description, [63](#)
 enhanced lists, [64](#)
 group lists, [63](#)
 personal lists, [63](#)
 system lists, [64](#)
 end-user procedures, [67](#)
 interactions, [68](#)
 screens, [65](#)
 Abbreviated Dialing List, [65](#)
 Optional Features, [64](#)
 Station, [66](#)
 troubleshooting, [69](#)
Access Security Gateway (ASG), [71](#)
 administering, [71](#)
 disabling, [74](#)
 modifying for lost key, [75](#)
 monitoring History Log, [75](#)
 prerequisites, [72](#)
 restarting, [75](#)
 setting up, [72](#)
 commands
 change login, [72](#), [74](#), [75](#)
 change system-parameters customer-options, [72](#)
 change system-parameters security, [73](#)
 list asg-history, [75](#)
 remove login, [75](#)

Access Security Gateway (ASG), (continued)
 description, [71](#)
 hardware requirements, [71](#)
 interactions, [77](#)
 screens, [72](#)
 Access Security Gateway Session History, [76](#)
 Login Administration, [73](#)
 Optional Features, [72](#)
 Security-Related System Parameters, [74](#)
AD, *see* Abbreviated Dialing (AD)
Administered Connections, [83](#)
 administering, [86](#)
 setting up, [87](#)
 commands
 change administered-connection, [85](#)
 display status-administered connection, [84](#)
 set time, [89](#)
 description, [83](#)
 interactions, [88](#)
 screens, [86](#)
 Access Endpoint, [86](#)
 Administered Connection, [86](#)
 Attendant Console, [86](#)
 Class of Restriction, [86](#)
 Class of Service, [86](#)
 Data Modules, [86](#)
 Dial Plan Record, [86](#)
 DS1 Circuit Pack, [86](#)
 Station, [86](#)
 Trunk Group, [86](#)
Administrable Language Displays, [91](#)
 administering, [92](#)
 entering translations for user-defined
 language, [94](#)
 prerequisites, [92](#)
 setting display language, [93](#)
 Unicode display, [95](#)
 commands
 change attendant, [93](#), [94](#), [96](#)
 change display-messages, [94](#)
 description, [91](#)
 screens, [93](#)
 Attendant Console, [94](#)
 System Parameters Country-Options, [92](#)
 troubleshooting, [97](#)
Administration Change Notification, [79](#)
 administering, [80](#)
 initiating, [80](#)
 commands
 notify history, [80](#)
 description, [79](#)
 screens, [80](#)
 Administration Changes, [81](#)

Administration Without Hardware (AWOH), [99](#)

- administering, [100](#)
- assigning for hunt-group queue, [101](#)
- assigning to attendant console, [103](#)
- assigning to data module, [103](#)
- assigning to telephone, [102](#)
- commands
 - change attendant, [103](#)
 - change data-module, [103](#)
 - change hunt-group, [101](#)
 - change station, [102](#)
- description, [99](#)
 - association and disassociation, [99](#)
 - disassociated telephones, [99](#)
 - duplicate telephones, [99](#)
 - Personal Station Access (PSA), [99](#)
 - phantom extensions, [100](#)
 - physical characteristics of telephone, [99](#)
 - Terminal Translation Initialization (TTI), [99](#)
 - user activated features, [99](#)
- interactions, [105](#)
- screens, [101](#)
 - Attendant Console, [103](#)
 - Data Module, [104](#)
 - Hunt Group, [101](#)
 - Station, [102](#)

AFRL, *see* Facility Restriction Levels, description, Alternate Facility Restriction Levels (AFRL)

Alphanumeric Dialing, [113](#)

- administering, [113](#)
- considerations, [114](#)
- description, [113](#)
- screens, [114](#)
 - Alphanumeric Dial Table, [114](#)

Announcements, [115](#)

- administering, [120](#)
 - adding announcement data modules, [123](#)
 - adding announcement extensions, [123](#)
 - converting announcement files to VAL format, [129](#)
 - converting for Avaya Interactive Voice Response (IVR), [130](#)
 - copying announcements, [126](#)
 - deleting and erasing announcements, [127](#)
 - deleting VAL announcements, [131](#)
 - displaying all announcements, [122](#)
 - managing VAL announcements using FTP, [132](#)
 - recording announcements, [124](#)
 - recording VAL announcements, [128](#)
 - restoring announcements, [126](#)
 - saving announcements, [126](#)
 - setting up continuous-play announcements, [127](#)
 - using SAT to delete all VAL announcements on circuit pack, [131](#)

Announcements, (continued)

- commands
 - add data-module, [123](#)
 - busyout board board-location, [131](#)
 - change announcements, [123](#), [127](#), [128](#), [135](#), [136](#)
 - copy announcements, [126](#)
 - delete filename.wav, [135](#)
 - disable filesystem board board-location, [137](#)
 - enable filesystem board, [133](#)
 - erase announcements, [127](#)
 - erase announcements board board-location, [132](#)
 - ftp val-ip-address, [133](#)
 - get filename.wav, [136](#)
 - list directory board, [131](#), [132](#)
 - list integrated-annc-boards, [122](#), [132](#)
 - put filename.wav, [136](#)
 - remove file board board-location /annc/ filename.wav, [131](#)
 - restore announcements, [126](#)
 - save announcements, [126](#)
- considerations, [138](#)
- description, [115](#)
 - announcement sources for G700 Media Gateway, [116](#)
 - barging-in on announcements, [119](#)
 - integrated announcements, [117](#)
 - local announcements on G700 Media Gateway, [117](#)
 - VAL announcements, [116](#)
 - VAL manager, [117](#)
- hardware requirements, [119](#)
- interactions, [139](#)
- Interactive Voice Response (IVR), [130](#)
- reports, [138](#)
- screens, [120](#)
 - Announcements/Audio Sources, [123](#)
 - Data Modules, [123](#)
 - Integrated Announcements, [122](#)
 - troubleshooting, [140](#)

Answer Detection, *see* Call Detail Recording

ARS, *see* Automatic Route Selection (ARS)

ASG, *see* Access Security Gateway (ASG)

Attendant Auto Start and Don't Split, [143](#)

- administering, [144](#)
 - assigning Don't Split button, [144](#)
 - prerequisites, [144](#)
- commands
 - change attendant, [144](#)
- considerations, [145](#)
- description, [143](#)
- hardware requirements, [143](#)
- interactions, [146](#)
- screens, [144](#)
 - Attendant Console, [145](#)

- Attendant Auto-Manual Splitting, [141](#)
 - administering, [141](#)
 - description, [141](#)
 - screens, [142](#)
 - Attendant Console, [142](#)
- Attendant Backup, [147](#)
 - administering, [148](#)
 - assigning console permissions to backup telephones, [151](#)
 - configuring your system, [149](#)
 - defining Class of Service console permissions, [151](#)
 - prerequisites, [148](#)
 - setting up telephones, [149](#)
- commands
 - change console-parameters, [149](#)
 - change cos, [151](#)
 - change station, [151](#)
- considerations, [154](#)
- description, [147](#)
- end-user procedures, [153](#)
- interactions, [154](#)
- screens, [149](#)
 - Class of Service, [148](#), [151](#)
 - Console Parameters, [150](#)
 - Feature Access Code (FAC), [149](#)
 - Station, [152](#)
- Attendant Call Waiting, [155](#)
 - administering, [156](#)
 - modifying timed intervals, [158](#)
 - prerequisites, [156](#)
 - setting up single-line telephones, [157](#)
- commands
 - change console-parameters, [158](#)
 - change station, [157](#)
 - change system-parameters features, [158](#)
- considerations, [159](#)
- description, [155](#)
- interactions, [160](#)
- screens, [156](#)
 - Feature-Related System Parameters, [158](#)
 - Station, [157](#)
- Attendant Calling of Inward Restricted Stations, [161](#)
 - administering, [161](#)
 - prerequisites, [162](#)
 - setting up Class of Restriction override, [162](#)
- description, [161](#)
- hardware requirements, [161](#)
- screens, [162](#)
- Attendant Conference, [165](#)
 - administering, [165](#)
 - prerequisites, [165](#)
 - setting up, [166](#)
- commands
 - change system-parameters features, [166](#)
- considerations, [167](#)
- description, [165](#)
- interactions, [168](#)
- screens, [166](#)
 - Feature-Related System-Parameters, [166](#)
- Attendant Control of Trunk Group Access, [169](#)
 - administering, [170](#)
 - assigning access buttons, [172](#)
 - prerequisites, [170](#)
 - setting trunk group threshold, [171](#)
- commands
 - change attendant, [172](#)
 - change trunk-group, [171](#)
- description, [169](#)
- hardware requirements, [170](#)
- interactions, [173](#)
- screens, [171](#)
 - Attendant Console, [172](#)
 - Trunk Group, [171](#)
- Attendant Direct Extension Selection (DXS), [175](#)
 - administering, [177](#)
 - prerequisites, [177](#)
 - considerations, [178](#)
 - description, [175](#)
 - Enhanced DXS Tracking, [176](#)
 - Group Display button, [176](#)
 - Standard DXS Tracking, [176](#)
 - hardware requirements, [176](#)
 - interactions, [178](#)
 - screens, [177](#)
- Attendant Direct Trunk Group Selection, [179](#)
 - administering, [180](#)
 - prerequisites, [180](#)
 - considerations, [181](#)
 - description, [179](#)
 - hardware requirements, [179](#)
 - interactions, [181](#)
 - screens, [180](#)
- Attendant Intrusion, [183](#)
 - administering, [183](#)
 - assigning an intrusion button, [184](#)
 - prerequisites, [183](#)
- commands
 - change attendant, [184](#)
- description, [183](#)
- hardware requirements, [183](#)
- interactions, [185](#)
- screens, [184](#)
 - Attendant Console, [184](#)
- Attendant Lockout – Privacy, [187](#)
 - administering, [187](#)
 - activating or deactivating, [188](#)
 - prerequisites, [188](#)
- commands
 - change console-parameters, [188](#)
- description, [187](#)
- hardware requirements, [187](#)
- interactions, [190](#)
- screens, [188](#)
 - Console Parameters, [189](#)
- Attendant Override of Diversion Features, [191](#)
 - administering, [191](#)
 - prerequisites, [191](#)
 - considerations, [192](#)
 - description, [191](#)
 - hardware requirements, [191](#)
 - interactions, [192](#)
 - screens, [192](#)

- Attendant Priority Queue, [193](#)
 - administering, [195](#)
 - assigning Call Type button, [198](#)
 - prerequisites, [195](#)
 - setting category priorities, [196](#)
 - setting number of calls in queue, [197](#)
 - commands
 - change console-parameters, [196](#)
 - change display-messages miscellaneous-features, [198](#)
 - change system-parameters features, [197](#)
 - considerations, [199](#)
 - description, [193](#)
 - hardware requirements, [195](#)
 - interactions, [199](#)
 - priority by call category, [193](#)
 - priority by call type, [194](#)
 - screens, [196](#)
 - Console Parameters, [196](#)
 - Feature-Related System Parameters, [197](#)
 - Language Translations, [198](#)
- Attendant Recall, [201](#)
 - administering, [201](#)
 - description, [201](#)
 - end-user procedures, [202](#)
 - interactions, [202](#)
 - screens, [202](#)
 - Attendant Console, [202](#)
- Attendant Serial Calling, [203](#)
 - administering, [203](#)
 - prerequisites, [203](#)
 - considerations, [204](#)
 - description, [203](#)
 - hardware requirements, [203](#)
 - interactions, [204](#)
 - screens, [204](#)
- Attendant Split Swap, [205](#)
 - administering, [205](#)
 - assigning split-swap button, [206](#)
 - prerequisites, [205](#)
 - commands
 - change attendant, [206](#)
 - description, [205](#)
 - hardware requirements, [205](#)
 - screens, [206](#)
 - Attendant Console, [206](#)
- Attendant Timers, [209](#)
 - administering, [210](#)
 - prerequisites, [210](#)
 - setting up, [211](#)
 - commands
 - change console-parameters, [211](#)
 - description, [209](#)
 - hardware requirements, [210](#)
 - interactions, [213](#)
 - screens, [211](#)
 - Console Parameters, [211](#)
- Attendant Trunk Identification, [215](#)
 - administering, [215](#)
 - prerequisites, [215](#)
 - considerations, [216](#)
 - description, [215](#)
 - hardware requirements, [215](#)
 - interactions, [216](#)
 - screens, [216](#)
- Attendant Vectoring, [221](#)
 - administering, [221](#)
 - assigning VDN extension to console, [224](#)
 - assigning VDN extension to tenant, [225](#)
 - creating VDN extension, [223](#)
 - prerequisites, [222](#)
 - commands
 - change console-parameters, [224](#)
 - change tenant, [225](#)
 - change vdn, [223](#)
 - change vector, [222](#)
 - display system-parameters customer-options, [222](#)
 - considerations, [226](#)
 - description, [221](#)
 - hardware requirements, [221](#)
 - interactions, [226](#)
 - screens, [223](#)
 - Call Vector, [222](#)
 - Console Parameters, [225](#)
 - Optional Features, [222](#)
 - Tenant, [225](#)
 - Vector Directory Number, [224](#)
- Audible Message Waiting, [217](#)
 - administering, [218](#)
 - for user, [218](#)
 - prerequisites, [218](#)
 - commands
 - change station, [218](#)
 - display system-parameters customer-options, [218](#)
 - considerations, [219](#)
 - description, [217](#)
 - screens, [218](#)
 - Optional Features, [218](#)
 - Station, [219](#)
- AUDIX One-Step Recording, [227](#)
 - administering, [229](#)
 - assigning feature button to telephone, [234](#)
 - assigning parameters, [230](#)
 - changing characteristics of zip tone, [234](#)
 - prerequisites, [229](#)
 - translating button label to user-defined language, [233](#)
 - translating feature display to user-defined language, [232](#)

- AUDIX One-Step Recording, (continued)
 - commands
 - change display-messages button-labels, [233](#)
 - change display-messages view-buttons, [232](#)
 - change station, [234](#)
 - change system-parameters country-options, [236](#)
 - change system-parameters features, [230](#)
 - change tone-generation, [235](#)
 - display system-parameters customer-options, [229](#)
 - considerations, [238](#)
 - description, [227](#)
 - feature button, [227](#)
 - language options, [227](#)
 - periodic alerting tone, [228](#)
 - ready indication tone, [228](#)
 - recording delay timer, [228](#)
 - end-user procedures, [237](#)
 - hardware requirements, [229](#)
 - hunt group extension number, [227](#)
 - initiator, [228](#), [231](#), [237](#)
 - interactions, [239](#)
 - screens, [230](#)
 - Feature-Related System Parameters, [231](#)
 - Language Translations, [232](#), [233](#)
 - Optional Features, [229](#), [799](#)
 - Station, [234](#)
 - System Parameters Country-Options, [236](#)
 - Tone Generation Customized Tones, [235](#)
 - troubleshooting, [240](#)
 - zip tone, [228](#), [234](#)
- Authorization Codes, [269](#)
 - administering, [271](#)
 - creating with specific COR, [274](#)
 - prerequisites, [271](#)
 - setting up, [272](#)
 - commands
 - change authorization-code, [273](#), [274](#)
 - change system-parameters features, [272](#)
 - display system-parameters customer-options, [271](#)
 - considerations, [275](#)
 - description, [269](#)
 - AAR and ARS calls, [270](#)
 - length of codes, [269](#)
 - using codes, [270](#)
 - interactions, [276](#)
 - Listed Directory Numbers (LDN), [275](#)
 - Remote Access, [955](#)
 - Remote Access Numbers (RAN), [275](#)
 - screens, [271](#)
 - Authorization Code - COR Mapping, [273](#), [274](#)
 - Feature-Related System Parameters, [272](#)
 - Optional Features, [271](#)
- Automated Attendant, [243](#)
 - administering, [243](#)
 - announcements, [245](#)
 - assigning caller information button on attendant consoles, [247](#)
 - assigning caller information button on multiappearance telephones, [246](#)
 - call vectors, [246](#)
 - controlling hunt groups by vector, [246](#)
 - prerequisites, [244](#)
 - setting the prompting timeout, [245](#)
 - vector directory number (VDN), [245](#)
 - commands
 - change attendant, [247](#)
 - change hunt-group, [246](#)
 - change station, [246](#)
 - change system-parameters features, [245](#)
 - display system-parameters customer-options, [244](#)
 - considerations, [248](#)
 - description, [243](#)
 - hardware requirements, [243](#)
 - interactions, [249](#)
 - screens, [244](#)
 - Announcements/Audio Sources, [244](#)
 - Attendant Console, [248](#)
 - Feature-Related System Parameters, [245](#)
 - Hunt Group, [246](#)
 - Optional Features, [244](#)
 - Station, [247](#)
 - Vector Directory Number, [244](#)
- Automatic Alternate Routing (AAR), [1109](#), [1165](#)
- Automatic Callback, [251](#)
 - administering, [253](#)
 - assigning Feature Access Code (FAC), [254](#)
 - assigning feature button, [255](#)
 - setting no-answer timeout interval, [254](#)
 - setting queue length for Ringback Queing, [256](#)
 - commands
 - add station, [255](#)
 - change feature-access-codes, [254](#)
 - change station, [255](#)
 - change system-parameters features, [254](#)
 - change trunk-group, [256](#)
 - considerations, [258](#)
 - description, [251](#)
 - Analog Busy Automatic Callback Without Flash, [252](#)
 - Ringback Queing, [252](#)
 - interactions, [259](#)
 - screens, [253](#)
 - Feature Access Code (FAC), [254](#)
 - Feature-Related System Parameters, [255](#)
 - Station, [256](#)
 - Trunk Group, [257](#)

Automatic Circuit Assurance (ACA), [261](#)
 administering, [262](#)
 description, [261](#)
 audit trail, [262](#)
 referral calls, [261](#)
 interactions, [263](#)
 reports, [263](#)
 screens, [262](#)
 Automatic Exclusion, see Privacy
 Automatic Number Identification, [265](#)
 description, [265](#)
 Outgoing Automatic Number Identification, [265](#)
 interactions, [266](#)
 Automatic Route Selection (ARS), [1110](#), [1165](#)
 AWOH, see Administration Without Hardware (AWOH)

B

Bridged Call Appearance, [277](#)
 administering, [279](#)
 assigning ringing to each appearance, [281](#)
 creating on multiappearance telephone, [281](#)
 creating on single-line telephone, [280](#)
 prerequisites, [279](#)
 commands
 change coverage path, [279](#)
 change station, [280](#), [281](#)
 display system-parameters features, [279](#)
 considerations, [283](#)
 description, [277](#)
 multiappearance telephone, [277](#)
 sample uses, [278](#)
 single-line telephone, [277](#)
 interactions, [284](#)
 screens, [279](#)
 Coverage Path, [279](#)
 Feature-Related System Parameters, [279](#)
 Station, [280](#) to [282](#)
 Bulletin Board, [293](#)
 administering, [294](#)
 changing bulletin board information, [295](#)
 prerequisites, [294](#)
 commands
 change bulletin-board, [295](#)
 change permissions, [294](#)
 considerations, [298](#)
 description, [293](#)
 capacity, [293](#)
 high priority messages, [293](#)
 screens
 Bulletin Board, [295](#)
 Command Permission Categories, [294](#)
 Busy Indicator, [299](#)
 description, [299](#)
 Busy Tone Disconnect, [299](#)
 interactions, [300](#)

Busy Verification, [301](#)
 administering, [303](#)
 prerequisites, [304](#)
 using Busy Verification, [304](#)
 considerations, [305](#)
 description, [301](#)
 interactions, [305](#)
 screens, [304](#)
 Station, [304](#)

C

Call Charge Information, [319](#)
 administering, [321](#)
 administering the charge display, [327](#)
 prerequisites, [321](#)
 commands
 change cor, [325](#)
 change system-parameters features, [323](#)
 considerations, [339](#)
 description, [319](#)
 Advice of Charge (AOC), [319](#)
 charge displays, [320](#)
 Periodic Pulse Metering (PPM), [319](#)
 end-user procedures, [338](#)
 interactions, [339](#)
 screens, [326](#)
 Feature-Related System Parameters, [323](#)
 viewing, [320](#)
 Call Coverage, [343](#)
 administering, [353](#)
 assigning coverage paths, [358](#), [360](#)
 assigning internal alerting, [366](#)
 assigning telephone numbers for off-network coverage, [361](#)
 assigning time-of-day coverage, [363](#)
 creating coverage answer groups, [365](#)
 creating coverage paths, [355](#)
 defining calls redirected off-network, [361](#)
 prerequisites, [353](#), [355](#), [358](#), [361](#), [363](#)
 commands
 add coverage answer-group, [365](#)
 add coverage path, [356](#)
 add coverage time-of-day, [363](#)
 change coverage path, [362](#)
 change coverage remote, [361](#)
 change feature-access-codes, [353](#)
 change station, [358](#), [360](#), [364](#)
 change trunk-group, [366](#)
 display coverage sender group, [356](#)
 display system-parameters call coverage/call forwarding, [355](#)
 display system-parameters customer-options, [361](#)
 considerations, [367](#)
 Coverage Subsequent Redirection interval, [344](#)

Call Coverage, (continued)

- description, [343](#)
 - announcement in coverage path, [349](#)
 - answer groups, [349](#)
 - caller response interval, [350](#)
 - changeable coverage paths, [347](#)
 - conditions that override, [351](#)
 - Consult, [350](#)
 - coverage criteria, [347](#)
 - coverage path, [344](#)
 - Extended User Administration of Redirected Calls, [347](#)
 - features that override, [350](#)
 - hunt group in coverage path, [349](#)
 - ISDN calls redirected off-net, [346](#)
 - multiple coverage paths, [345](#)
 - notifying users upon redirection, [349](#)
 - off-network coverage, [346](#)
 - redirection, [347](#), [348](#), [351](#)
 - subsequent redirection interval, [349](#)
 - Time-of-Day Coverage, [345](#)
 - VDN in Coverage Path (VICP), [348](#)
 - Directed Call Pickup, [351](#)
 - hardware requirements, [352](#)
 - interactions, [367](#)
 - reports, [366](#)
 - screens, [354](#)
 - Coverage Answer Group, [365](#)
 - Coverage Path, [356](#), [362](#)
 - Feature Access Code (FAC), [354](#)
 - Optional Features, [361](#)
 - Remote Call Coverage Table, [361](#)
 - Station, [358](#), [364](#)
 - System-Parameters Call Coverage/Call Forwarding, [355](#)
 - Time of Day Coverage Table, [363](#)
 - Trunk Group, [366](#)
 - Switch Communication Interface (SCI) link, [348](#)
 - troubleshooting, [370](#)
- Call Detail Recording, [373](#)
- administering, [421](#)
 - for data module, [440](#)
 - for paging ports, [450](#)
 - identifying Inter-Exchange Carrier, [448](#)
 - intra-switch CDR, [450](#)
 - port field values, [445](#)
 - prerequisites, [421](#), [424](#)
 - system parameters, [428](#)
 - to trunk group, [437](#)
 - commands
 - change cor, [425](#)
 - change data-module, [440](#)
 - change feature-access-codes, [421](#)
 - change intra-switch-cdr, [450](#)
 - change ixc-codes, [448](#)
 - change loudspeaker paging, [450](#)
 - change station, [425](#), [427](#)
 - change system-parameters cdr, [427](#), [428](#)
 - change trunk-group, [426](#), [437](#)
 - display system-parameters customer-options, [422](#)

Call Detail Recording, (continued)

- considerations, [452](#)
- customized call record formats, [385](#)
- data record formats
 - 59 character, [388](#)
 - enhanced expanded, [399](#)
 - enhanced LSU, [396](#)
 - enhanced printer, [392](#)
 - enhanced TELESEER, [387](#)
 - enhanced unformatted, [402](#)
 - expanded, [397](#)
 - int process, [403](#)
 - int-direct, [404](#)
 - int-ISDN, [405](#)
 - ISDN LSU, [395](#)
 - ISDN printer, [390](#)
 - ISDN-TELESEER, [386](#)
 - LSU, [394](#)
 - LSU, LSU-expand, unformatted, and customized, [383](#), [384](#)
 - LSU-expand, [393](#)
 - printer, [389](#)
 - printer and expanded, [384](#)
 - TELESEER 59 character, int-proc, int-direct, and int-ISDN, [384](#)
 - unformatted, [401](#)
- description, [373](#)
 - Account Code Dialing, [375](#)
 - answer detection, [374](#)
 - Answer Supervision by Timeout, [375](#)
 - attendant transfer of an incoming trunk call, [381](#)
 - call classification, [374](#)
 - call splitting, [376](#)
 - CDR output, [383](#)
 - CDR Privacy, [382](#)
 - CDR record formats, [383](#)
 - date record formats, [383](#)
 - Forced Entry of Account Codes (FEAC), [375](#)
 - incoming trunk call splitting, [376](#)
 - intraswitch CDR, [381](#)
 - ITCS and OTCS and attendant call recording, [380](#)
 - ITCS conference call on the same server, [377](#)
 - ITCS transfer to an outgoing trunk, [378](#)
 - monitor call detail records, [374](#)
 - Network Answer Supervision, [375](#)
 - OTCS call transfer, [380](#)
 - OTCS conference call, [379](#)
 - outgoing trunk call splitting, [379](#)
- end-user procedures, [451](#)
- examples
 - attendant call transfer on a public-network trunk, [381](#)
 - attendant incoming trunk call transfer, [380](#)
 - ITCS and a call transfer on the same server, [377](#)
 - ITCS and a call transfer to the public network, [378](#)
 - ITCS and a conference call, [377](#)
 - OTCS and a call transfer to a public network, [379](#)
 - OTCS and a conference call on the public network, [379](#)

Call Detail Recording, (continued)

- interactions, [452](#)
- screens, [423](#)
 - CDR System Parameters, [427](#), [429](#), [436](#)
 - Class of Restriction, [425](#)
 - Data Module, [440](#), [441](#)
 - Feature Access Code (FAC), [422](#)
 - Inter-Exchange Carrier Codes, [448](#), [449](#)
 - Intra-Switch CDR, [451](#)
 - Loudspeaker Paging, [450](#)
 - Optional Features, [422](#)
 - Station, [426](#), [428](#)
 - Trunk Group, [426](#), [438](#)

Call Forwarding, [463](#)

- administering, [466](#)
 - assigning All Calls, [469](#)
 - assigning Busy/Don't Answer, [470](#)
 - assigning Off-Net, [472](#)
 - disabling Override, [474](#)
 - enabling Override, [474](#)
 - prerequisites, [467](#), [470](#), [472](#)
 - removing Busy/Don't Answer, [471](#)
 - removing Off-Net, [473](#)
 - viewing active users, [469](#)

commands

- change feature-access-codes, [467](#)
- change station, [469](#) to [471](#), [473](#)
- change system-parameters coverage-forwarding, [470](#) to [474](#)
- display station, [469](#)
- display system-parameters customer-options, [472](#)
- list call-forwarding, [469](#)
- save translation, [477](#)

considerations, [477](#)Coverage of Calls Redirected Off-Net (CCRON), [465](#)description, [463](#)

- attendants, [464](#)
- Busy/Don't Answer, [464](#)
- Call Forwarding All Calls, [463](#)
- Call Forwarding Off-Net, [465](#)
- Call Forwarding Override, [465](#)
- coverage for unanswered forwarded calls, [466](#)
- Feature Access Code (FAC), [464](#)
- security, [466](#)
- user notification, [466](#)

end-user procedures, [475](#)hardware requirements, [466](#)interactions, [477](#)off-network timer, [465](#)screens, [469](#)

- Feature Access Code (FAC), [467](#)
- Optional Features, [472](#)
- Station, [470](#)
- System Parameters Call Coverage/Call Forwarding, [468](#), [471](#) to [474](#)

V1, V2, or V3 system, [469](#)

Call Offer, see Attendant Intrusion

Call Park, [481](#)

- administering, [482](#)
 - assigning call park button to multiple-call appearance telephone, [486](#)
 - prerequisites, [482](#)
- commands
 - change console-parameters, [485](#)
 - change feature-access-codes, [482](#)
 - change station, [486](#)
 - change system-parameters features, [483](#)
- considerations, [489](#)
- description, [481](#)
- end-user procedures, [487](#)
- interactions, [489](#)
- screens, [486](#)
 - Console-Parameters, [485](#)
 - Feature Access Code (FAC), [483](#)
 - Feature-Related System Parameters, [484](#)
 - Station, [487](#)

Call Pickup, [491](#)

- administering, [493](#)
 - assigning call pickup button, [494](#)
 - assigning Directed Call Pickup, [498](#)
 - assigning extended pickup group to another extended pickup group, [504](#)
 - assigning pickup group to extended pickup group, [501](#)
 - assigning user to pickup group, [496](#)
 - changing call pickup button, [496](#)
 - prerequisites, [494](#), [498](#), [501](#), [504](#)
 - removing Directed Call Pickup, [501](#)
 - removing extended pickup group from another extended pickup group, [507](#)
 - removing pickup group from extended pickup group, [504](#)
 - removing user from pickup group, [498](#)

call appearance selection, [492](#)call distribution algorithms, [492](#)

commands

- add pickup-group, [496](#)
- change cor, [499](#)
- change extended-pickup-group, [503](#), [504](#), [506](#), [507](#)
- change feature-access-codes, [494](#), [502](#), [505](#)
- change pickup-group, [497](#), [498](#)
- change station, [495](#), [496](#), [500](#), [501](#)
- change system-parameters features, [498](#), [502](#), [505](#)

considerations, [508](#)description, [491](#)

- call ringing, [491](#)
- Directed Call Pickup, [492](#)
- Extended Group Pickup, [492](#)
- status lamp, [491](#)

end-user procedures, [508](#)interactions, [508](#)

- Call Pickup, (continued)
 - screens, [493](#)
 - Class of Restriction, [500](#)
 - Extended Pickup Group, [504](#), [507](#)
 - Feature Access Code (FAC), [494](#), [503](#), [506](#)
 - Feature-Related System Parameters, [499](#), [502](#), [505](#)
 - Station, [495](#), [501](#)
- Call Waiting Termination, [511](#)
 - administering, [512](#)
 - assigning call waiting termination, [513](#)
 - prerequisites, [512](#)
 - commands
 - change station, [513](#)
 - change system-parameters features, [512](#)
 - considerations, [515](#)
 - description, [511](#)
 - call waiting tones, [511](#)
 - hardware requirements, [511](#)
 - interactions, [515](#)
 - screens, [513](#)
 - Feature-Related System Parameters, [512](#)
 - Station, [514](#)
- Call Waiting, see Call Waiting Termination
- Call-by-Call Service Selection, [307](#)
 - administering, [311](#)
 - administering network facilities, [316](#)
 - prerequisites, [312](#)
 - setting up a trunk group for CBC, [313](#)
- Call Detail Recording, [310](#)
 - commands
 - add trunk-group, [313](#)
 - change isdn network-facilities, [316](#)
 - change route-pattern, [315](#)
 - change trunk-group, [313](#)
 - display system-parameters customer-options, [312](#)
 - considerations, [318](#)
 - description, [307](#)
 - incoming call-handling treatment, [310](#)
 - interactions, [318](#)
 - ISDN messages and information elements for usage allocation, [308](#)
 - screens, [312](#)
 - CBC Trunk Group Usage Allocation, [314](#)
 - CBC Trunk Group Usage Allocation Plan Assignment Schedule, [315](#)
 - Incoming Call Handling Treatment, [314](#)
 - ISDN Trunk Group, [313](#)
 - Route Pattern, [316](#)
 - Usage Allocation Plans, [309](#)
- Caller ID, [517](#)
 - administering, [518](#)
 - displaying ICLID Information, [519](#)
 - prerequisites, [518](#)
 - commands
 - add trunk group, [519](#)
 - change trunk group, [519](#)
 - display system-parameters customer-options, [518](#)
 - considerations, [521](#)
- Caller ID, (continued)
 - description, [517](#)
 - interactions, [521](#)
 - screens, [518](#)
 - Trunk Group, [519](#)
 - Trunk Group Administrable Timers, [520](#)
 - Trunk Group Trunk Features, [520](#)
- CAS, see Centralized Attendant Service (CAS)
- CDR, see Call Detail Recording
- Centralized Attendant Service (CAS), [523](#)
 - administering, [525](#)
 - prerequisites, [525](#)
 - commands
 - display system-parameters customer-options, [525](#)
 - considerations, [526](#)
 - description, [523](#)
 - hardware requirements, [525](#)
 - interactions, [527](#)
 - listed directory number (LDN), [523](#)
 - release link trunks (RLT), [523](#)
 - screens, [525](#)
 - Optional Features, [525](#)
- Class of Restriction (COR), [531](#)
 - administering, [535](#)
 - allowing users to change, [540](#)
 - displaying administered CORs, [536](#)
 - prerequisites, [541](#)
 - setting up, [537](#)
 - commands
 - change cor, [537](#)
 - change feature-access-codes, [541](#)
 - change system-parameters features, [542](#)
 - display system-parameters customer-options, [541](#)
 - list cor, [536](#)
 - description, [531](#)
 - called party restrictions, [533](#)
 - calling party restrictions, [532](#)
 - COR-to-COR restrictions, [535](#)
 - fully restricted service, [535](#)
 - strategy for assigning, [532](#)
 - end-user procedures, [542](#)
 - Facility Restriction Level (FRL), [532](#)
 - interactions, [543](#)
 - screens, [536](#)
 - Class of Restriction, [533](#), [535](#), [538](#) to [540](#)
 - Class of Restriction Information, [537](#)
 - Feature Access Code (FAC), [541](#)
 - Feature-Related System Parameters, [542](#)
 - Optional Features, [541](#)
 - Trunk Group, [533](#)
- Class of Service, [547](#)
 - administering, [547](#)
 - assigning a COS, [551](#)
 - defining COS for your system, [548](#)
 - descriptions of the COS features, [549](#)
 - commands
 - change cos, [548](#)
 - considerations, [552](#)
 - description, [547](#)

Class of Service, (continued)
 interactions, [552](#)
 screens, [548](#)
 Class of Service, [549](#)
 commands, *see* commands *under individual feature names*
 Conference, [561](#)
 administering, [564](#)
 assigning a toggle-mute feature button, [567](#)
 assigning the conf-dsp, fe-mute, and no-hold-conf feature buttons, [568](#)
 prerequisites, [564](#)
 commands
 change attendant, [570](#)
 change station, [567](#), [569](#)
 change system-parameters features, [564](#)
 display system-parameters customer-options, [568](#)
 considerations, [572](#)
 description, [561](#)
 No Dial Tone Conferencing, [562](#)
 No Hold Conference, [562](#)
 Selective Conference Party Display, Drop, and Mute, [563](#)
 Transfer Toggle/Swap, [562](#)
 end-user procedures, [571](#)
 displaying participants on call, [571](#)
 hardware requirements, [563](#)
 interactions, [573](#)
 number of participants on a call, [561](#)
 screens, [566](#)
 Attendant Console, [570](#)
 Feature-Related System Parameters, [564](#)
 Optional Features, [568](#)
 Station, [567](#), [569](#)
 conference tones, [998](#)
 considerations, *see* considerations *under individual feature names*; *see also* interactions *under individual feature names*
 Consult, *see* Call Coverage
 COR, *see* Class of Restriction (COR)
 Co-Resident DEFINITY Local Area Network Gateway, [555](#)
 administering, [556](#)
 prerequisites, [557](#)
 commands
 add data-module, [557](#)
 add ip-interface, [557](#)
 change node-names ip, [557](#)
 display ip-interfaces, [557](#)
 display node-names ip, [557](#)
 display system-parameters customer-options, [557](#)
 list data-modules, [557](#)
 description, [555](#)
 hardware requirements, [556](#)
 screens, [558](#)
 COS, *see* Class of Service
 Crisis Alert, *see* Enhanced 911 (E911)

D

Data Call Setup, [575](#)
 administering, [578](#)
 assigning data extension feature button, [590](#)
 defining data module, [580](#)
 prerequisites, [578](#)
 specifying port location, [590](#)
 commands
 add data-module, [580](#)
 change feature-access-codes, [578](#)
 change modem-pool, [590](#)
 change station, [590](#)
 considerations, [594](#)
 description, [575](#)
 call progress messages, [576](#)
 cause code, [576](#)
 reason code, [576](#)
 special characters, [575](#)
 end-user procedures, [591](#)
 interactions, [594](#)
 screens, [579](#)
 Data Module, [580](#), [581](#)
 Feature Access Code (FAC), [579](#)
 Modem Pool Group, [590](#)
 Station, [591](#)
 Data Privacy, *see* Privacy
 Data Restriction, *see* Privacy
 Default Dialing, [597](#)
 administering, [597](#)
 description, [597](#)
 screens, [598](#)
 Demand Print, [599](#)
 administering, [599](#)
 description, [599](#)
 hardware requirements, [599](#)
 screens, [599](#)
 Dial Access to Attendant, [601](#)
 administering, [601](#)
 changing attendant access code, [602](#)
 commands
 change dialplan analysis, [602](#)
 description, [601](#)
 interactions, [603](#)
 screens, [602](#)
 Dial Plan Analysis Table, [602](#)
 Dial Plan, [605](#)
 administering, [609](#)
 adding extension ranges, [611](#)
 adding Feature Access Codes, [611](#)
 displaying your dial plan, [609](#), [610](#)
 modifying your dial plan, [610](#)
 prerequisites, [609](#)
 commands
 change dialplan analysis, [610](#), [611](#)
 display dialplan analysis, [610](#)
 display system-parameters customer-options, [609](#)

Dial Plan, (continued)
 considerations, [612](#)
 description, [605](#)
 Dial Plan Analysis Table, [605](#)
 Dial Plan Parameters, [606](#)
 Multi-Location Dial Plan, [607](#)
 other options, [608](#)
 Uniform Dial Plan (UDP), [608](#)
 interactions, [612](#)
 screens, [609](#)
 Dial Plan Analysis Table, [605](#), [610](#)
 Dial Plan Parameters, [607](#)
 Feature Related System Parameters, [612](#)
 Optional Features, [609](#)
 Trunk Group, [606](#)
 Distinctive Ringing, [615](#)
 administering, [615](#)
 defining, [616](#)
 commands
 change system-parameters features, [616](#)
 considerations, [618](#)
 description, [615](#)
 Personalized Ringing, [615](#)
 interactions, [618](#)
 screens, [616](#)
 Feature-Related System Parameters, [617](#)
 Dual Homing, see Multiple Level Precedence and Pre-emption (MLPP)
 DXS, see Attendant Direct Extension Selection (DXS)

E

E911, see Enhanced 911
 End Office Access Line Hunting, see Multiple Level Precedence and Preemption (MLPP)
 end-user procedures, see end-user procedures *under individual feature names*
 Enhanced 911 (E911), [619](#)
 administering, [625](#)
 prerequisites, [626](#), [627](#), [629](#), [631](#)
 setting up CAMA numbering, [634](#)
 setting up Crisis Alert to attendant or display telephone, [627](#)
 setting up Crisis Alert to notify digital pager, [631](#)
 setting up emergency extension forwarding, [633](#)
 Automatic Location Information (ALI) database, [620](#)
 Automatic Number Identification (ANI), [636](#)
 Avaya IP Softphone, [620](#)
 Caller Emergency Service Identification (CESID) number, [620](#)
 Calling Party Number (CPN), [619](#)
 Centralized Automatic Message Accounting (CAMA) trunks, [619](#)

Enhanced 911 (E911), (continued)
 commands
 change ars analysis location, [621](#)
 change attendant, [629](#)
 change cama-numbering, [634](#)
 change station, [628](#), [629](#), [630](#)
 change system-parameters crisis-alert, [630](#), [631](#)
 change system-parameters features, [633](#)
 display system-parameters customer-options, [627](#)
 list emergency, [625](#)
 considerations, [636](#)
 Crisis Alert, [624](#)
 description, [619](#)
 Location Specific Routing, [621](#)
 media gateways in different locations, [620](#)
 Emergency Location Information Number (ELIN), [622](#)
 hardware requirements, [625](#)
 Integrated Services Digital Network (ISDN) trunks, [619](#)
 interactions, [637](#)
 public safety answering point (PSAP), [619](#)
 reports, [635](#)
 screens, [626](#)
 ARS Digit Analysis Table, [628](#)
 Attendant Console, [629](#), [630](#)
 CAMA Numbering - E911 Format, [634](#)
 Class of Restriction, [626](#)
 Crisis Alert System Parameters, [630](#), [632](#)
 Feature Access Code (FAC), [626](#)
 Feature-Related System Parameters, [633](#)
 Optional Features, [627](#)
 Route Pattern, [626](#)
 Station, [620](#), [629](#)
 Universal Emergency Number (UEN), [619](#)
 wired IP telephones, [622](#)
 enhanced tip ring (ETR) telephone, [908](#)
 Extended User Administration of Redirected Calls, [639](#)
 administering
 assigning a station security code to a user, [646](#)
 prerequisites, [642](#)
 commands
 add telecommuting-access, [642](#)
 change feature-access-codes, [642](#)
 change station, [646](#)
 description, [639](#)
 COR, [640](#)
 COS, [640](#)
 DCS, [640](#)
 disabling the telecommuting access extension, [639](#)
 off-site locations, [640](#)
 end-user procedures, [647](#)
 interactions, [649](#)
 screens, [645](#)
 Feature Access Code (FAC), [643](#)
 Station, [647](#)
 Telecommuting Access, [642](#)

F

- FAC, *see* Feature Access Codes (FAC)
- Facility and Non-Facility Associated Signaling, [651](#)
- administering, [653](#)
 - implementing FAS and NFAS, [653](#)
 - reviewing the guidelines for coordinating FAS and NFAS, [653](#)
- description, [651](#)
- D-Channel backup activation, [652](#)
 - D-Channel backup with NFAS, [651](#)
- screens, [653](#)
- DS1 Circuit Pack, [655](#)
 - Interface Links, [653](#)
 - ISDN-PRI Trunk Group, [657](#)
 - Processor Channel, [654](#)
 - Signaling Group, [656](#)
- Facility Restriction Levels, [659](#)
- administering, [661](#)
 - considerations, [663](#)
 - description, [659](#)
 - AAR and ARS calls, [659](#)
 - Alternate Facility Restriction Levels (AFRL), [660](#)
 - alt-fri feature button, [661](#)
 - authorization codes, [661](#)
 - end-user procedures, [663](#)
 - interactions, [664](#)
 - screens, [662](#)
 - toll fraud prevention, [660](#)
- Facility Restrictions Levels
- Traveling Class Marks (TCM), [659](#)
- FEAC, *see* Call Detail Recording, Forced Entry of Account Codes (FEAC)
- Feature Access Codes (FAC), [665](#)
- administering, [666](#)
 - assigning, [666](#)
 - changing or deleting, [667](#)
 - prerequisites, [666](#)
 - analog rotary dial telephones, [665](#)
 - commands
 - change feature-access-codes, [666](#), [667](#)
 - description, [665](#)
 - * and #, [665](#)
 - access codes, [665](#)
 - activate and deactivate codes, [665](#)
 - lock and unlock codes, [665](#)
 - send and cancel codes, [665](#)
 - screens, [666](#)
 - Feature Access Code (FAC), [667](#), [668](#)
 - troubleshooting, [669](#)
- FRL, *see* Facility Restriction Levels

G

- Group Paging, [671](#)
- administering, [672](#)
 - changing paging group, [673](#)
 - creating paging group, [672](#)
 - viewing all paging groups, [674](#)
 - commands
 - add group-page, [672](#)
 - change group-page, [673](#)
 - list group-page, [674](#)
 - description, [671](#)
 - hardware requirements, [671](#)
 - interactions, [674](#)
 - screens, [672](#)
 - Group Paging Using Speakerphone, [673](#)
 - Speakerphone Page Groups, [674](#)
 - troubleshooting, [676](#)

H

- H0 channels, [1159](#)
- H11 channels, [1158](#)
- H12 channels, [1158](#)
- hardware requirements, *see* hardware requirements *under individual feature names*
- Hold, [677](#)
- administering, [678](#)
 - assigning FAC for CAS remote hold and answer, [679](#)
 - enabling Automatic Hold, [679](#)
 - prerequisites, [678](#)
 - commands
 - change feature-access-codes, [679](#)
 - change system-parameters features, [679](#)
 - considerations, [681](#)
 - description, [677](#)
 - Automatic Hold, [677](#)
 - Hard Hold, [677](#)
 - multiappearance telephones, [677](#)
 - single-line telephones, [677](#)
 - Soft Hold, [677](#)
 - interactions, [681](#)
 - screens, [678](#)
 - Feature Access Code (FAC), [680](#)
 - Feature-Related System Parameters, [679](#)
- Hot Line Service, [683](#)
- administering, [684](#)
 - considerations, [685](#)
 - description, [683](#)
 - interactions, [685](#)
 - screens, [684](#)

Hunt Groups, [687](#)
 administering, [693](#)
 adding announcements, [697](#)
 changing, [696](#)
 setting up groups, [694](#)
 setting up night service, [698](#)
 setting up queues, [696](#)
 commands
 add hunt-group, [694](#)
 change hunt-group, [696](#) to [698](#)
 display announcements, [697](#)
 considerations, [699](#)
 description, [688](#)
 announcements, [688](#)
 call coverage, [689](#)
 call distribution methods, [689](#)
 Call Forwarding All Calls, [692](#)
 delay announcement intervals, [688](#)
 extension unavailability, [691](#)
 Hunt Group Busy option, [691](#)
 queues, [692](#)
 Send All Calls, [692](#)
 TTY, [692](#)
 interactions, [700](#)
 screens, [694](#)
 Hunt Group, [695](#), [697](#), [698](#)
 Hunt Group (for group member assignments), [696](#)

I

IAS, see Inter-PBX Attendant Service (IAS)
 Individual Attendant Access, [703](#)
 administering, [703](#)
 assigning extension to attendant console, [704](#)
 commands
 add attendant, [704](#)
 description, [703](#)
 screens, [704](#)
 Attendant Console, [704](#)
 interactions, see interactions under individual feature names; see also considerations under individual feature names
 Intercom, [711](#)
 administering, [712](#)
 description, [711](#)
 groups, [711](#)
 telephones, [712](#)
 interactions, [713](#)
 screens, [712](#)
 Internal Automatic Answer, [715](#)
 Internal Automatic Answer (IAA)
 administering, [716](#)
 considerations, [717](#)
 description, [715](#)
 feature operations, [716](#)
 interactions, [717](#)
 screens, [716](#)

Inter-PBX Attendant Service
 commands
 change console-parameters, [708](#)
 display system-parameters customer-options, [708](#)
 screens
 Console Parameters, [709](#)
 Optional Features, [708](#)
 Inter-PBX Attendant Service (IAS), [707](#)
 administering, [707](#)
 enabling Inter-PBX Attendant Service, [708](#)
 prerequisites, [708](#)
 description, [707](#)
 hardware requirements, [707](#)
 interactions, [710](#)
 screens, [708](#)
 ISDN Service, [721](#)
 administering, [731](#)
 description, [721](#)
 ISDN interworking, [726](#)
 national ISDN-2 services, [725](#)
 private network services, [724](#)
 transmission rate and protocols, [722](#)
 interactions, [732](#)
 screens, [731](#)
 IVR, see Announcements, Interactive Voice Response (IVR)

L

Last Number Dialed, [735](#)
 administering, [735](#)
 considerations, [736](#)
 description, [735](#)
 interactions, [736](#)
 screens, [735](#)
 Attendant Console, [735](#)
 Feature Access Code (FAC), [735](#)
 Station, [735](#)
 LDN, see Listed Directory Number (LDN)
 Leave Word Calling (LWC), [737](#)
 considerations, [739](#)
 description, [737](#)
 voice synthesis, [738](#)
 interactions, [739](#)
 Line Load Control (LLC), see Multiple Level Precedence and Preemption (MLPP)
 Line Lockout, [741](#)
 administering, [742](#)
 considerations, [743](#)
 description, [741](#)
 screens, [742](#)
 Listed Directory Number (LDN), [745](#)
 administering, [746](#)
 assigning incoming destination to trunk, [747](#)
 assigning listed directory numbers, [746](#)
 commands
 change listed-directory-number, [746](#)
 change trunk-group, [747](#)
 considerations, [749](#)

Listed Directory Number (LDN), (continued)
 description, [745](#)
 routing incoming DID trunk calls to attendant groups, [745](#)
 routing incoming FX and CO trunk calls to attendant groups, [745](#)
 interactions, [749](#)
 screens, [746](#)
 Listed Directory Numbers, [747](#)
 Trunk Group, [748](#)

Loss Plans, [751](#)
 administering, [752](#)
 prerequisites, [752](#)
 commands
 display location-parameters, [752](#)
 display system-parameters customer-options, [752](#)
 considerations, [753](#)
 description, [751](#)
 interactions, [753](#)
 screens, [752](#)
 Location Parameters, [752](#)
 Optional Features, [752](#)

Loudspeaker Paging, [755](#)
 administering, [757](#)
 prerequisites, [757](#)
 setting up chime paging over loudspeakers, [760](#)
 setting up voice paging over loudspeakers, [758](#)
 description, [755](#)
 chime paging, [756](#)
 deluxe paging, [755](#)
 restrictions, [756](#)
 interactions, [762](#)
 screens, [758](#)
 troubleshooting, [764](#)

M

Magic-on-Hold system, see Tenant Partitioning

Malicious Call Trace (MCT), [765](#)
 considerations, [768](#)
 description, [765](#)
 activating, [765](#)
 controlling, [766](#)
 deactivating, [766](#)
 ISDN notification, [767](#)
 using voice recoder, [766](#)
 interactions, [769](#)
 screens, [767](#)

Manual Exclusion, see Privacy

Manual Message Waiting, [771](#)
 administering, [771](#)
 assigning feature button, [772](#)
 commands
 change station, [772](#)
 description, [771](#)
 screens, [772](#)
 Station, [772](#)

Manual Originating Line Service, see Hot Line Service

Manual Signaling, [775](#)
 administering, [775](#)
 description, [775](#)
 end-user procedures, [776](#)
 interactions, [776](#)
 screens, [775](#)
 Station, [775](#)

Meet-me Conference, [777](#)
 administering, [777](#)
 accessing as attendee, [784](#)
 changing access code, [784](#)
 creating or changing vector, [778](#)
 creating vector directory number (VDN), [782](#)
 prerequisites, [778](#)
 using Selective Conference Party Display, Drop, and Mute, [785](#)
 commands
 add vdn, [782](#)
 change vector, [778](#)
 display system-parameters customer-options, [778](#)
 list meet-me vdn, [783](#)
 considerations, [786](#)
 description, [777](#)
 end-user procedures, [784](#)
 hardware requirements, [777](#)
 interactions, [788](#)
 screens, [778](#)
 Call Vector, [779](#), [780](#)
 Meet-me Vector Directory Number, [783](#)
 Optional Features, [778](#)
 Vector Directory Number, [782](#)
 troubleshooting, [789](#)

MLPP, see Multiple Level Precedence and Preemption (MLPP)

Modem Pooling, [791](#)
 administering, [792](#)
 considerations, [793](#)
 description, [791](#)
 interactions, [793](#)
 screens, [792](#)

Multifrequency Signaling, [807](#)
 considerations, [810](#)
 description, [807](#)
 guidelines, [808](#)
 MF Shuttle, [808](#)
 MFE, [808](#)
 R2 multifrequency compelled (MFC)
 signaling, [808](#)
 interactions, [811](#)
 screens, [810](#)

Multi-Location Dial Plan, [795](#)
 administering, [799](#)
 changing extensions, [800](#)
 prepending numbers, [802](#)
 prerequisites, [799](#)
 setting up announcements, [803](#)
 setting up local centralized answering point (LCAP), [803](#)
 setting up multiple FACs – ARS, [804](#)
 setting up multiple FACs for attendants, [804](#)

Multi-Location Dial Plan, (continued)
 commands
 change daylight-savings-rules, [799](#)
 display system-parameters customer-
 options, [799](#)
 description, [796](#)
 announcements, [797](#)
 feature access codes (FACs), [797](#)
 location prefix, [796](#)
 maintenance, [798](#)
 distributed communication system (DCS), [795](#)
 interactions, [805](#)
 local centralized answering point (LCAP), [803](#)
 screens, [799](#)
 Change Station Extension, [800](#)
 Daylight Savings Rules, [799](#)
 Uniform Dial Plan Table, [802](#)
Multiple Level Precedence and Preemption (MLPP), [815](#)
 administering, [831](#)
 adding extensions, [840](#)
 announcements for Precedence Calling, [840](#)
 assigning announcement types, [841](#)
 assigning attendant queue priorities, [839](#), [852](#)
 assigning digit analysis, [845](#)
 assigning digit conversion, [847](#)
 assigning feature access codes, [844](#)
 assigning hot line number, [838](#), [851](#)
 assigning LLC level to COR, [849](#)
 assigning LLC level to system, [849](#)
 assigning maximum precedence levels, [836](#)
 assigning MLPP feature access code, [834](#)
 assigning Precedence Calling system
 parameters, [834](#)
 assigning Preemption to COR, [849](#)
 assigning route patterns, [847](#)
 assigning trunks, [836](#), [849](#)
 assigning WNDP feature access codes, [851](#)
 assigning WNDP system parameters, [850](#)
 deleting announcements, [842](#)
 Dual Homing, [848](#)
 enabling Precedence Call Waiting, [843](#)
 End Office Access Line Hunting, [848](#)
 Line Load Control (LLC), [849](#)
 Precedence Call Waiting, [843](#)
 Precedence Calling, [834](#)
 Precedence Routing, [844](#)
 Preemption, [849](#)
 prerequisites, [832](#)
 recording announcements, [841](#)
 saving announcements, [843](#)
 setting Precedence Call timeout, [843](#), [849](#)
 Worldwide Numbering and Dialing Plan
 (WNDP), [850](#)
 Attendant Diversion Timing, [817](#)
 Call Purpose Indicators, [818](#)
 circuit pack updates, [830](#)

Multiple Level Precedence and Preemption (MLPP),
(continued)
 commands
 add station, [838](#), [852](#)
 add trunk-group, [836](#)
 change announcements, [840](#)
 change console-parameters, [839](#)
 change cor, [836](#), [849](#)
 change feature-access-codes, [834](#), [844](#), [851](#)
 change precedence-routing analysis, [845](#)
 change precedence-routing digit-conversion, [848](#)
 change route-pattern, [847](#)
 change station, [843](#)
 change system-parameters mlpp, [834](#), [841](#), [843](#),
 [849](#), [850](#)
 display system-parameters customer-
 options, [832](#)
 considerations, [852](#)
 Default Route Digit, [829](#)
 Defense Switched Network (DSN), [815](#)
 description, [815](#)
 announcements for Precedence Calling, [824](#)
 Dual Homing, [825](#)
 End Office Access Line Hunting, [826](#)
 Line Load Control (LLC), [826](#)
 Precedence Call Waiting, [825](#)
 Precedence Calling, [816](#)
 Precedence Routing, [825](#)
 Preemption, [826](#)
 Worldwide Numbering and Dialing Plan
 (WNDP), [828](#)
 Generic Switching Center Requirements
 (GSCR), [815](#)
 hardware requirements, [830](#)
 interactions, [854](#)
 Joint Interoperability Test Command (JITC), [815](#)
 Line Load Control (LLC), [849](#)
 lockdown, [826](#)
 Precedence Call Timeout, [818](#)
 Precedence calling tones, [817](#)
 Remote Attendant Route String, [817](#)
 Route Control Digit, [854](#)
 screens, [832](#)
 Abbreviated Dialing List, [839](#), [852](#)
 Announcements/Audio Sources, [840](#)
 Class of Restriction, [836](#), [849](#)
 Console Parameters, [839](#)
 Feature Access Code (FAC), [834](#), [844](#), [851](#)
 Multiple Level Precedence & Preemption
 Parameters, [834](#), [841](#), [843](#), [849](#), [850](#)
 Optional Features, [832](#)
 Pattern Number, [847](#)
 Precedence Routing Digit Analysis Table, [845](#)
 Precedence Routing Digit Conversion Table, [848](#)
 Station, [838](#), [843](#), [852](#)
 Trunk Features, [836](#)
 WNDP Emergency 911 Route String, [862](#)
 Multiple Music-on-Hold, see Tenant Partitioning

Music-on-Hold, [863](#)
 administering, [864](#)
 assigning music source port, [867](#)
 assigning music tones, music ports, and music for transferred trunks, [865](#)
 connecting music source to server, [867](#)
 defining COR, [866](#)
 American Society of Composers, Artists, and Producers (ASCAP), [863](#)
 Broadcast Music Incorporated (BMI), [863](#)
 commands
 add trunk-group, [867](#)
 change cor, [866](#)
 change music-sources, [867](#)
 change system-parameters features, [865](#)
 considerations, [869](#)
 description, [863](#)
 copyrighted material, [863](#)
 options for music, silence, and tone, [863](#)
 interactions, [869](#)
 screens, [864](#)
 Class of Restriction, [867](#)
 Feature-Related System Parameters, [865](#)
 Music Sources, [868](#)

N

N x DS0 channels, [1159](#)
 Night Service, [871](#)
 administering, [873](#)
 setting up external alerting, [879](#)
 setting up external alerting Night Service, [880](#)
 setting up night console service, [876](#)
 setting up Night Service for hunt groups, [882](#)
 setting up Night Service for trunk groups, [882](#)
 setting up night station service, [877](#)
 setting up night station service to voice mail, [874](#)
 setting up trunk answer from any station, [879](#)
 commands
 add hunt group, [874](#)
 change attendant, [876](#)
 change console-parameters, [875](#), [878](#), [879](#), [881](#)
 change feature-access-codes, [879](#)
 change hunt-group, [882](#)
 change listed-directory-numbers, [875](#), [877](#), [880](#)
 change tenant, [875](#)
 change trunk-group, [882](#)
 considerations, [883](#)
 description, [871](#)
 night console service, [871](#)
 night station service, [871](#)
 Trunk Answer from Any Station (TAAS), [872](#)
 Trunk Groups, [872](#)
 interactions, [885](#)

Night Service, (continued)
 screens, [873](#)
 Attendant Console, [877](#)
 Console Parameters, [876](#), [878](#), [880](#), [881](#)
 Feature Access Code (FAC), [879](#)
 Hunt Group, [874](#), [883](#)
 Listed Directory Number, [878](#), [880](#)
 Listed Directory Numbers, [875](#)
 Trunk Group, [882](#)

O

Off-Premises Station, [889](#)
 administering, [890](#)
 activating for user, [890](#)
 commands
 change station, [890](#)
 description, [889](#)
 interactions, [892](#)
 screens, [890](#)
 Station, [891](#)

P

paging, see Group Paging
 Periodic Pulse Metering (PPM), [319](#)
 Personal Station Access (PSA), [893](#)
 administering, [895](#)
 description, [893](#)
 button mapping, [894](#)
 dissociated telephones, [894](#)
 invalid attempts using, [893](#)
 preferences and permissions, [893](#)
 security, [894](#)
 telecommuting, [893](#)
 unanswered calls, [894](#)
 end-user procedures, [896](#)
 interactions, [899](#)
 screens, [895](#)
 Personalized Ringing, [901](#)
 administering, [902](#)
 assigning to user telephone, [902](#)
 commands
 change station, [902](#)
 description, [901](#)
 power failures, [901](#)
 ringing patterns, [901](#)
 interactions, [904](#)
 screens, [902](#)
 Station, [903](#)

- Posted Messages, [905](#)
 - *, [920](#), [922](#)
 - administering, [908](#)
 - activating the Posted Messages security code, [912](#)
 - creating custom messages, [915](#)
 - defining feature access code (FAC), [910](#)
 - prerequisites, [908](#)
 - sending custom messages through QSIG, [913](#)
 - setting up a telephone security code, [911](#)
 - translating button label to user-defined language, [918](#)
 - translating feature button to user-defined language, [917](#)
 - translating softkey button label to user-defined language, [917](#)
 - translating system messages to user-defined language, [914](#)
 - attendant consoles, [920](#)
 - commands
 - change display-messages button-labels, [918](#)
 - change display-messages posted-messages, [914](#), [915](#)
 - change display-messages softkey-labels, [917](#)
 - change display-messages view-buttons, [917](#)
 - change feature-access-codes, [910](#), [913](#)
 - change station, [911](#)
 - change system-parameters features, [912](#)
 - display system-parameters customer-options, [908](#)
 - considerations, [923](#)
 - description, [905](#)
 - custom messages, [907](#)
 - fixed messages, [906](#)
 - language options, [905](#)
 - QSIG support, [907](#)
 - end-user procedures, [919](#)
 - enhanced tip ring (ETR) telephone, [908](#)
 - hardware requirements, [908](#)
 - interactions, [924](#)
 - message posting mode, [906](#), [923](#), [925](#)
 - screens, [909](#)
 - Custom Posted Messages, [916](#)
 - Feature Access Code (FAC), [911](#)
 - Feature-Related System Parameters, [913](#), [914](#)
 - Language Translations, [917](#) to [919](#)
 - Optional Features, [908](#)
 - Station, [912](#)
 - System Posted Messages, [915](#)
 - security code, [911](#), [920](#)
 - selection display mode, [906](#), [923](#), [925](#)
 - special dial tone, [906](#)
- PPM, see Periodic Pulse Metering (PPM)
- Precedence Call Waiting, see Multiple Level Precedence and Preemption (MLPP)
- Precedence Calling, see Multiple Level Precedence and Preemption (MLPP)
- Precedence Routing, see Multiple Level Precedence and Preemption (MLPP)
- Preemption, see Multiple Level Precedence and Preemption (MLPP)
- Priority Calling, [927](#)
 - administering, [927](#)
 - assigning priority feature button to attendant console, [930](#)
 - assigning priority feature button to telephone, [931](#)
 - Class of Service, [929](#)
 - Feature Access Code, [929](#)
 - prerequisites, [928](#)
 - commands
 - change attendant, [930](#)
 - change feature-access-codes, [929](#)
 - change station, [931](#)
 - change system-parameters features, [928](#)
 - description, [927](#)
 - end-user procedures, [931](#)
 - interactions, [932](#)
 - screens, [929](#)
 - Attendant Console, [930](#)
 - Feature Access Code (FAC), [929](#)
 - Feature-Related System Parameters, [928](#)
 - Station, [931](#)
- Privacy, [935](#)
 - administering, [936](#)
 - activating Data Restriction for trunk group, [939](#)
 - for user, [937](#)
 - prerequisites, [936](#)
 - commands
 - change cos, [936](#)
 - change feature-access-codes, [936](#)
 - change station, [937](#)
 - change system-parameters features, [936](#)
 - change trunk-group, [939](#)
 - considerations, [940](#)
 - description, [935](#)
 - Data Privacy, [935](#)
 - Data Restriction, [935](#)
 - Privacy - Automatic Exclusion, [935](#)
 - Privacy - Manual Exclusion, [936](#)
 - end-user procedures, [940](#)
 - interactions, [941](#)
 - screens, [937](#)
 - Class of Service, [936](#)
 - Feature Access Codes (FAC), [936](#)
 - Feature-Related System Parameters, [936](#)
 - Station, [937](#), [938](#)
 - Trunk Group, [939](#)
- PSA, see Personal Station Access (PSA)
- Public Network Call Priority, [943](#)
 - administering, [945](#)
 - description, [943](#)
 - China, [943](#)
 - Russia, [944](#)
 - Spain, [945](#)
 - interactions, [946](#)
 - China, [947](#)
 - Russia, [948](#)
 - screens, [945](#)

Q

QSIG

- Call Forwarding, [479](#)
- definition, [907](#)
- Multi-Location Dial Plan, [795](#)
- Posted Messages, [907](#) to [914](#)
- World Class Routing, [1170](#)

R

Recorded Telephone Dictation Access, [951](#)

- description, [951](#)
- interactions, [952](#)

Remote Access, [953](#)

- administering, [956](#)
 - administering remote access, [957](#)
 - administering Remote Access for a trunk group, [963](#)
 - assigning authorization codes, [960](#)
 - disabling Remote Access, [959](#)
 - enabling Remote Access, [957](#)

AFRLs, [956](#)authorization codes, [955](#)

commands

- change authorization-codes, [962](#)
- change remote-access, [957](#), [959](#)
- change system-parameters features, [960](#)
- change trunk-group, [963](#)
- display capacity, [962](#)
- display system-parameters customer-options, [960](#)
- status remote-access, [954](#), [955](#)

considerations, [964](#)description, [953](#)

- alternate facility restriction levels, [956](#)
- authorization codes, [955](#)
- barrier codes, [953](#), [955](#)
- COR, [956](#)
- logoff notification, [956](#)
- Night Service, [954](#)
- security, [953](#), [954](#)

end-user procedures, [963](#)interactions, [964](#)screens, [957](#)

- Authorization Code - COR Mapping, [962](#)
- Feature-Related System Parameters, [960](#)
- Optional Features, [960](#)
- Remote Access, [958](#), [959](#)
- Security-Related System Parameters, [959](#)
- System Copyright, [959](#)
- Trunk Group, [963](#)
- security, [953](#), [961](#), [962](#)

reports

- Announcements, [138](#)
- Automatic Circuit Assurance (ACA), [263](#)
- Call Coverage, [366](#)
- Enhanced 911 (E911), [635](#)
- Security Violation Notification (SVN), [983](#)
- Station Hunting, [1010](#)
- Uniform Dial Plan (UDP), [1135](#)

Restriction - Controlled, [965](#)

- administering, [966](#)
- considerations, [967](#)
- description, [965](#)
- end-user procedures, [966](#)
- interactions, [967](#)
- screens, [966](#)

Ringing - Abbreviated and Delayed, [969](#)

- administering, [970](#)
 - assigning abbreviated ringing button to user, [973](#)
 - assigning ring control to user, [972](#)
 - prerequisites, [970](#)

commands

- change station, [972](#), [973](#)
- change system-parameters features, [970](#)

considerations, [974](#)description, [969](#)

- ringing types, [969](#)

interactions, [974](#)screens, [971](#)

- Feature-Related System Parameters, [971](#)

- Station, [972](#), [973](#)

RNX, see Uniform Dial Plan (UDP), network location code (RNX)

RTL, see Centralized Attendant Service (CAS), release link trunks (RLT)

S

screens, *see* screens under individual feature names

security

- Remote Access, [953](#), [961](#), [962](#)
- Terminal Translation Initialization, [1089](#)
- Security Violation Notification (SVN), [977](#)

administering, [980](#)

- prerequisites, [980](#)
- setting up, [981](#)

commands

- change remote-access, [982](#)
- change station, [982](#)
- change system-parameters security, [981](#)
- clear measurements security-violations, [978](#)
- disable login, [979](#)
- disable remote-access, [979](#)
- display svn-button-location, [978](#)
- display system-parameters customer-options, [980](#)
- enable login, [979](#)
- enable remote-access, [979](#)
- monitor security-violations, [978](#)

Security Violation Notification (SVN), (continued)
 considerations, [984](#)
 description, [977](#)
 dealing with security violations, [979](#)
 disable login IDs, [979](#)
 disabling remote access, [979](#)
 enabling login IDs, [979](#)
 enabling remote access, [979](#)
 event sequence, [978](#)
 reporting, [978](#)
 security violation thresholds and notification, [977](#)
 SVN - halt buttons, [978](#)
 SVN referral call with announcement, [979](#)
 interactions, [984](#)
 reports, [983](#)
 screens, [980](#)
 Remote Access, [982](#)
 Security Related System Parameters, [981](#)
 Station, [983](#)
 Separation of Bearer and Signaling, [985](#)
 administering, [985](#)
 considerations, [986](#)
 description, [985](#)
 interactions, [987](#)
 screens, [986](#)
 Service Observing, [997](#)
 administering, [999](#)
 description, [997](#)
 listen and talk modes, [998](#)
 restrictions, [998](#)
 telephone displays, [998](#)
 trunk calls, [998](#)
 warning and conference tones, [998](#)
 end-user procedures, [1000](#)
 interactions, [1001](#)
 screens, [999](#)
 speed dialing, see Abbreviated Dialing (AD)
 Station Hunting, [1005](#)
 administering, [1007](#)
 assigning hunt-to station to extension, [1008](#)
 assigning Station Hunting after Coverage, [1007](#)
 enabling before Coverage, [1009](#)
 commands
 change coverage-path, [1007](#)
 change station, [1008](#)
 change system-parameters coverage-forwarding, [1009](#)
 description, [1005](#)
 duplicating station, [1006](#)
 removing station, [1006](#)
 routing calls through Station Hunting chain, [1005](#)
 Station Hunting and Call Coverage, [1006](#)
 hardware requirements, [1006](#)
 interactions, [1011](#)
 reports, [1010](#)
 screens, [1007](#)
 Coverage Path, [1008](#)
 Station, [1009](#)
 System Parameters Call Coverage/Call Forwarding, [1010](#)

Station Lock, [1015](#)
 administering, [1016](#)
 description, [1015](#)
 end-user procedures, [1016](#)
 interactions, [1017](#)
 screens, [1016](#)
 COR, [1016](#)
 Feature Access Code (FAC), [1016](#)
 Station, [1016](#)
 Station Security Code, [1019](#)
 administering, [1019](#)
 creating station security code, [1020](#)
 commands
 change feature-access-codes, [1020](#)
 change station, [1020](#)
 change system-parameters security, [1020](#)
 description, [1019](#)
 interactions, [1021](#)
 screens, [1019](#)
 Feature Access Codes (FAC), [1020](#)
 Security-Related System Parameters, [1020](#)
 Station, [1020](#)
 Supporting TTY Callers, [1023](#)
 administering, [1026](#)
 description, [1023](#)
 handling TTY calls with vectors, [1025](#)
 setting up announcements for TTY callers, [1024](#)
 setting up hunt groups for TTY callers, [1024](#)
 SVN, see Security Violation Notification (SVN)
 system security
 disabling login IDs, [979](#)
 disabling remote access, [979](#)
 enabling login IDs, [979](#)
 enabling remote access, [979](#)

T

TAAS, see Night Service, description, Trunk Answer from Any Station (TAAS)
 TCM, see Facility Restriction Levels, Traveling Class Marks (TCM)
 Telephone Display, [1027](#)
 administering, [1055](#)
 description, [1027](#)
 interactions, [1056](#)
 screens, [1056](#)
 Date/Time Mode and Formats – English, [1035](#)
 Date/Time Mode and Formats – French, Italian, Spanish, and User-Defined, [1035](#)
 Language Translations, [1037](#)
 Leave Word Calling Format – English, [1039](#)
 Leave Word Calling Formats – French, Italian, Spanish, and User-Defined, [1039](#)
 Temporary Bridged Appearance, [1083](#)
 administering, [1084](#)
 considerations, [1084](#)
 description, [1083](#)
 interactions, [1085](#)
 screens, [1084](#)

- Tenant Partitioning, [1059](#)
 - administering, [1063](#)
 - assigning tenant partition number to attendant, [1069](#)
 - assigning sources of music for tenant partitions, [1076](#)
 - assigning tenant partition number to access telephone, [1067](#)
 - assigning tenant partition number to agent login ID, [1068](#)
 - assigning tenant partition number to announcement, [1069](#)
 - assigning tenant partition number to data module, [1070](#)
 - assigning tenant partition number to hunt group, [1071](#)
 - assigning tenant partition number to loudspeaker paging zone, [1072](#)
 - assigning tenant partition number to remote access extension, [1073](#)
 - assigning tenant partition number to terminating extension group, [1074](#)
 - assigning tenant partition number to trunk group, [1075](#)
 - assigning tenant partition number to user extension, [1073](#)
 - assigning tenant partition number to vector directory number, [1075](#)
 - defining tenant partition, [1065](#)
 - prerequisites, [1064](#)
- commands
 - change access-endpoint, [1067](#)
 - change agent loginid, [1068](#)
 - change announcements, [1069](#)
 - change attendant, [1069](#)
 - change data-module, [1070](#)
 - change hunt-group, [1071](#)
 - change music-sources, [1076](#)
 - change paging loudspeaker, [1072](#)
 - change remote-access, [1073](#)
 - change station, [1073](#)
 - change tenant, [1065](#)
 - change term-ext-group, [1074](#)
 - change trunk-group, [1075](#)
 - change vdn, [1075](#)
 - display system-parameters customer-options, [1064](#)
- description, [1059](#)
 - access control, [1060](#)
 - attendant services, [1061](#)
 - examples, [1061](#)
 - Magic-on-Hold system, [1059](#)
 - Multiple Music-on-Hold, [1062](#)
 - network route selection, [1061](#)
 - Offer B, [1059](#)
 - partitioning tenants, [1060](#)
- Tenant Partitioning, (continued)
 - interactions, [1078](#)
 - screens, [1064](#)
 - Access Endpoint, [1068](#)
 - Agent LoginID, [1068](#)
 - Announcements/Audio Sources, [1069](#)
 - Attendant Console, [1070](#)
 - Data Module, [1071](#)
 - Hunt Group, [1072](#)
 - Loudspeaker Paging, [1072](#)
 - Music Sources, [1076](#)
 - Optional Features, [1064](#)
 - Remote Access, [1073](#)
 - Station, [1074](#)
 - Tenant, [1065](#)
 - Terminating Extension Group, [1074](#)
 - Trunk Group, [1075](#)
 - Vector Directory Number, [1076](#)
- Terminal Translation Initialization, [1087](#)
 - administering, [1089](#)
 - description, [1087](#)
 - analog queue warning ports and external alert ports, [1089](#)
 - attendant consoles, [1087](#)
 - data modules, [1088](#)
 - ISDN-BRI telephones, [1088](#)
 - security, [1089](#)
 - voice and data telephones, [1088](#)
 - screens, [1089](#)
- Terminating Extension Group, [1093](#)
 - administering, [1093](#)
 - considerations, [1094](#)
 - description, [1093](#)
 - interactions, [1094](#)
 - screens, [1094](#)
- Time-of-Day Routing, see Call Coverage
- Transfer, [1097](#)
 - administering, [1099](#)
 - considerations, [1100](#)
 - description, [1097](#)
 - abort transfer, [1097](#)
 - emergency transfer, [1099](#)
 - outgoing trunk to outgoing trunk transfer, [1098](#)
 - pull transfer, [1097](#)
 - transfer recall, [1098](#)
 - transfer upon hangup, [1098](#)
 - trunk-to-trunk transfer, [1098](#)
 - interactions, [1101](#)
 - screens, [1099](#)
- troubleshooting
 - Abbreviated Dialing (AD), [69](#)
 - Administrable Language Displays, [97](#)
 - Announcements, [140](#)
 - AUDIX One-Step Recording, [240](#)
 - Call Coverage, [370](#)
 - Feature Access Codes (FAC), [669](#)
 - Group Paging, [676](#)
 - Loudspeaker Paging, [764](#)
 - Meet-me Conference, [789](#)

Trunk Flash, [1105](#)
 administering, [1106](#)
 considerations, [1107](#)
 description, [1105](#)
 end-user procedures, [1106](#)
 screens, [1106](#)
 TTI, see Terminal Translation Initialization

U

UDP, see Uniform Dial Plan (UDP)
 Uniform Dial Plan (UDP), [1109](#)
 administering, [1114](#)
 BCC Value field entries, [1119](#)
 CA-TSC field entries, [1119](#)
 Hop Lmt field entries, [1120](#)
 Inserted Digits field entries, [1121](#)
 ISDN protocols, [1131](#)
 ITC field entries, [1121](#)
 IXC field entries, [1122](#)
 LAR field entries, [1122](#)
 Numbering Format field entries, [1123](#)
 Prefix Mark field entries, [1124](#)
 prerequisites, [1114](#)
 Service Feature field entries, [1125](#)
 setting up AAR Digit Analysis Table, [1129](#)
 setting up AAR Digit Conversion Table, [1126](#)
 setting up ARS Digit Analysis Table, [1132](#)
 setting up ARS Digit Conversion Table, [1128](#)
 setting up extension number portability numbering plan, [1134](#)
 setting up node number routing, [1126](#)
 setting up route pattern, [1117](#)
 setting up Uniform Dial Plan Table, [1116](#)
 commands
 add station, [1110](#)
 change aar analysis, [1129](#)
 change aar digit-conversion, [1126](#)
 change ars analysis, [1132](#)
 change ars digit-conversion, [1128](#)
 change enp-number-plan, [1134](#)
 change feature-access-codes, [1114](#)
 change node-number routing, [1126](#)
 change route-pattern, [1117](#)
 change uniform-dial plan, [1116](#)
 display system-parameters customer-options, [1114](#)
 considerations, [1135](#)

Uniform Dial Plan (UDP), (continued)
 description, [1109](#)
 AAR code extension treatment, [1110](#)
 administration and processing example, [1111](#)
 Automatic Alternate Routing (AAR), [1109](#)
 Automatic Route Selection (ARS), [1110](#)
 code extension treatment, [1110](#)
 ENP code extension treatment, [1110](#)
 extension, [1110](#)
 extension number conversion, [1109](#)
 Extension Number Portability (ENP), [1110](#)
 local extension treatment, [1110](#)
 matching pattern selections, [1111](#)
 RNX and UDP codes, [1111](#)
 temporary out of service extension treatment, [1110](#)
 Extended Trunk Access (ETA), [1111](#)
 interactions, [1136](#)
 network location code (RNX), [1111](#)
 reports, [1135](#)
 screens, [1115](#)
 AAR Digit Analysis Table, [1130](#)
 AAR Digit Conversion Table, [1127](#)
 ARS Digit Analysis Table, [1132](#)
 ARS Digit Conversion Table, [1128](#)
 Extension Number Portability Numbering Plan, [1135](#)
 Node Number Routing, [1126](#)
 Optional Features, [1114](#)
 Route Pattern, [1118](#)
 Uniform Dial Plan Table, [1116](#)

V

VIAS, see Visually Impaired Attendant Service (VIAS)
 Visually Impaired Attendant Service (VIAS), [1137](#)
 administering, [1138](#)
 prerequisites, [1138](#)
 description, [1137](#)
 hardware requirements, [1138](#)
 interactions, [1139](#)
 screens, [1138](#)
 Voice Message Retrieval, [1141](#)
 administering, [1141](#)
 description, [1141](#)
 interactions, [1143](#)
 screens, [1142](#)

W

warning tones, [998](#)

Whisper Paging, [1145](#)

administering, [1146](#)

prerequisites, [1146](#)

description, [1145](#)

call redirection overrides, [1145](#)

network restrictions, [1146](#)

speakerphones, [1146](#)

end-user procedures, [1147](#)

interactions, [1149](#)

screens, [1147](#)

Wideband Switching, [1151](#)

administering, [1162](#)

blocking prevention, [1161](#)

channel allocation, [1152](#)

considerations, [1163](#)

data backup connection, [1156](#)

description, [1151](#)

channel allocation, [1152](#)

endpoint applications with signaling, [1153](#)

guidelines and examples, [1155](#)

ISDN-PRI trunk groups and channel

allocation, [1157](#)

non-signaling endpoint applications, [1154](#)

typical uses, [1152](#)

direction of trunk/hunting within facilities, [1157](#)

facility lists, [1157](#)

glare prevention and resolution, [1160](#)

H0 channels, [1159](#)

H11 channels, [1158](#)

H12 channels, [1158](#)

high-speed video conferencing, [1155](#)

interactions, [1163](#)

N x DS0 channels, [1159](#)

nonsignaling configurations, [1154](#)

port network (PN) preferential trunk routing, [1157](#)

screens, [1162](#)

World Class Routing, [1165](#)

administering, [1171](#)

adding new area code or prefix, [1183](#)

assigning ARS FAC, [1175](#)

assigning telephone to partition group, [1188](#)

defining interexchange carrier calls, [1181](#)

defining local information calls, [1183](#)

defining operator-assisted calls, [1180](#)

displaying ARS analysis information, [1178](#)

managing calling privileges using COR and

FRL, [1173](#)

prerequisites, [1177](#), [1186](#), [1190](#)

setting up location ARS FAC, [1177](#)

setting up partition groups, [1187](#)

setting up Time of Day Routing, [1190](#)

using ARS to restrict outgoing calls, [1185](#)

using restricted area code prefixes, [1181](#)

using wildcards, [1182](#)

World Class Routing, (continued)

commands

change ars analysis, [1181](#) to [1185](#)

change cor, [1174](#), [1189](#)

change dialplan analysis, [1175](#)

change features-access-codes, [1176](#)

change locations, [1177](#)

change partition-route-table, [1187](#)

change station, [1174](#), [1190](#)

display ars analysis, [1178](#), [1180](#), [1181](#)

display system-parameters customer-

options, [1177](#), [1187](#), [1190](#)

display time-of-day, [1190](#)

display toll, [1185](#)

list ars analysis, [1179](#)

list ars route-chosen, [1183](#), [1187](#)

list cor, [1188](#)

description, [1165](#)

ARS analysis, [1167](#)

ARS dialing without FAC, [1169](#)

Automatic Alternate Routing (AAR), [1165](#)

Automatic Route Selection (ARS), [1165](#)

digit conversion, [1168](#)

extensions, [1170](#)

overview, [1166](#)

partitioning, [1171](#)

interactions, [1193](#)

interexchange carrier (IXC), [1181](#)

screens, [1172](#)

ARS Digit Analysis Table, [1179](#) to [1184](#), [1186](#)

ARS Route Chosen Report, [1184](#), [1187](#)

Class of Restriction, [1175](#), [1189](#)

Class of Restriction Information, [1189](#)

Dial Plan Analysis Table, [1176](#)

Feature Access Code (FAC), [1177](#)

Locations, [1178](#)

Optional Features, [1177](#), [1186](#), [1190](#)

Partition Routing Table, [1188](#)

Station, [1174](#), [1190](#)

Time of Day Routing Plan, [1191](#)

wildcards, [1181](#), [1182](#)

Worldwide Numbering and Dialing Plan (WNDP), see

Multiple Level Precedence and Preemption (MLPP)