Administration for Network Connectivity for Avaya Communication Manager

555-233-504
Issue 9.1
January 2005

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 Contacts* link. Then click the appropriate link for the type of support you need.
- Outside the United States, click the *Escalation Contacts* 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 using 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 Information Technology Equipment, AS/NZS 60950:2000.

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, EN55022:1998, and AS/NZS 3548.

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

CE

Avaya Inc. declares that the equipment specified in this document bearing the "CE" (*Conformité Europeénne*) mark conforms to the European Union Radio and Telecommunications Terminal Equipment Directive (1999/5/EC), including the Electromagnetic Compatibility Directive (89/336/EEC) and Low Voltage Directive (73/23/EEC).

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

About this document

Purpose

This document describes how to implement Voice over IP (VoIP) applications on IP and DCS networks through Avaya Communication Manager administration. It is intended primarily for persons involved in planning, designing, or administering VoIP networks. For installation or upgrade procedures between VoIP components or for connecting adjuncts/peripherals to a configuration, refer to the upgrades and installation documents for the respective equipment.

In addition to VoIP applications, considerable information is provided as well on the design and administration of:

- Distributed Communications System on page 163
- Extended Trunk Access on page 216
- Inter-PBX Attendant Service on page 219
- Centralized Voice Mail on page 226
- Japan TTC Q931-a on page 231

Content

The information in this book is presented as follows:

- Chapter 1: Networking overview provides an overview of network connectivity and IP addressing.
- Chapter 2: Administering converged networks provides procedures for initial administration of server-to-gateway connections, including a sample network configuration procedure with administration screens, IP trunks using H.323 IP connections, DCS AUDIX and CMS adjunct administration, installing and administering Avaya IP telephones, and administering IP-to-IP connections.
- Chapter 3: Network quality administration provides instructions for administering Quality of Service on telephony and network equipment.
- Chapter 4: Administering dedicated networks describes several types of private networks and related services.
- Chapter 5: Feature interactions and considerations describes the DCS, QSIG, and Italian TGU/TGE features and feature interactions.

- Appendix A: Using IP Routes describes when to use IP routes and how to administer them.
- Appendix B: Internet Control Message Protocol (ICMP) ECHO messages presents a current listing of when and why the IMCP pings are used, and the consequences of ping failures caused by real network outages or ICMP message filtering or suppression.
- Index

Note:

> "Chapter 5: Troubleshooting" in the June, 2004, issue has been removed and incorporated into *Maintenance Procedures for Avaya Communication Manager 2.2, Media Gateways and Servers*, 03-300192, Issue 3, January 2005.

Conventions used in this book

Terminology

- The word "system" is a general term encompassing all references to the Avaya servers running Avaya Communication Manager.
- Circuit pack codes (for example, TN780 or TN2182B) are shown with the *minimum acceptable* alphabetic suffix (like the "B" in the code TN2182B).

  Generally, an alphabetic suffix higher than that shown is also acceptable. However, not every *vintage* of either the minimum suffix or a higher suffix code is necessarily acceptable. A suffix of "P" means that firmware can be downloaded to that circuit pack.
- The term "ASAI" is synonymous with the newer CallVisor ASAI.
- UUCSS refers to a circuit pack address in cabinet-carrier-slot order.

Typographic

Other terms and conventions might help you use this book.

- Names or titles of screens, windows, or dialog boxes are printed in bold italic, as follows: ***screen_name***.
- Commands are printed in bold face as follows: **`command`**.

  We show complete commands in this book, but you can usually type an abbreviated version of the command. For example, **`list configuration station`** can be typed as **`list config sta`**.

- Command variables are printed in bold italics as follows: ***variable***.
- Screen displays are printed in constant width as follows: **screen display**.

  A screen is any form displayed on your computer or terminal monitor.
- Names of fields on screens or in dialog boxes are printed in bold face, as follows: **field_name**.
- Keys, buttons, mouse clicks, and other types of input are printed as follows: **Key.**
- To move to a certain field, you can use the **Tab** key, arrows, or the **Enter** key (the **Enter** key may appear as the **Return** key on your keyboard).
- If you use terminal emulation software, you need to determine what keys correspond to **ENTER**, **RETURN**, **CANCEL**, **HELP**, **NEXT PAGE**, etc.
- In this book we use the terms “telephone” and “voice terminal” to refer to phones.
- If you need help constructing a command or completing a field entry, remember to use **Help**.
  - When you press **Help** at any point on the command line, a list of available commands appears.
  - When you press **Help** with your cursor in a field on a screen, a list of valid entries for that field appears.
- The status line or message line can be found near the bottom of your monitor display. This is where the system displays messages for you. Check the message line to see how the system responds to your input. Write down the message if you need to call our helpline.
- When a procedure requires you to press **Enter** to save your changes, the screen you were working on clears and the cursor returns to the command prompt.

  The message line shows **command successfully completed** to indicate that the system accepted your changes.

Admonishments

Admonishments in this book have the following meanings:

⚠ **CAUTION:**

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

⚠ **WARNING:**

Denotes possible harm to hardware or equipment.

⚠ **DANGER:**

Denotes possible harm or injury to your body.

Physical dimensions

- Physical dimensions in this book are in inches (in.) followed by metric centimeters (cm) in parentheses.
- Wire gauge measurements follow the AWG standard followed by the cross-sectional area in millimeters squared (mm^2) in parentheses.

How to get this book

On the Web

If you have Internet access, you can view and download the latest version of this book. To view the book, you must have a copy of Acrobat Reader.

To access the latest version of this book

1. At your browser, go to the Avaya web site:

   http://www.avaya.com
2. Select **Support**.
3. Select **Product Documentation**.
4. Select **Communication Systems** from the left menu bar.
5. Scroll down to find the latest release of DEFINITY or Avaya Communication Manager documents.

Non-Web

This book and any other DEFINITY or Avaya Communication Manager books can be ordered directly from:

Globalware Solutions
200 Ward Hill Avenue
Haverhill, MA 01835 USA

Toll-free numbers

+1-800-457-1235 (phone)
+1-800-457-1764 (fax)

Non-800 numbers

+1 207-866-6701 (phone)
+1 207-626-7269 (fax)

How to get technical assistance

To get technical assistance and trouble escalation

1. At your browser, go to the Avaya web site:

   http://www.avaya.com

2. Select **Support**.
3. Select **1-800 Support Directory**.

- If you are within the United States
  - The *Avaya TSO Support* link lists toll-free numbers for various support services.
  - The *Escalation Lists* link contains additional links to regional service centers.
- If you are outside the United States, select click *Avaya Centers of Excellence for non-USA*, which includes phone numbers for the regional Centers of Excellence.

If you do not have Web access, use the phone numbers in Table 1: Avaya support on page 18.

Note:

You may need to purchase an extended service agreement to use some of these resources. See your Avaya representative for more information.

Table 1: Avaya support

Support	Number
● Avaya Technical Consulting - System Support (formerly known as the DEFINITY Helpline) for help with feature administration and system applications	+1-800-225-7585
● Avaya National Customer Care Center Support Line for help with maintenance and repair	+1-800-242-2121
● Avaya Toll Fraud Intervention	+1-800-643-2353
● Avaya Corporate Security	+1-877-993-8442
● International Technical Assistance Center (ITAC)	+905-943-8801
For all international resources, contact your local Avaya authorized dealer for any additional help and questions.	

Security

To ensure the greatest security possible for customers, Avaya offers services that can reduce toll-fraud liabilities. Contact your Avaya representative for more security information. Login security is an attribute of Communication Manager. Existing passwords expire 24 hours after installation.

Antistatic protection

⚠ **CAUTION:**

When handling circuit packs or any system components, always wear an antistatic wrist ground strap. Connect the strap to an approved ground such as an unpainted metal surface.

Remove/Install circuit packs

⚠ **CAUTION:**

When the power is on:

- *The control circuit packs cannot be removed or installed.*
- *The port circuit packs can be removed or installed.*

Standards compliance

The equipment in this document complies with the following standards (as applicable).

Environmental requirements and safety standards

Table 2: Regulatory Compliance

Standard	**Country/Region**
EN60950	Western Europe
UL 1950., ULC C22.2.950	USA and Canada
Global IEC, CB /Scheme Report IEC	Global
950	
AS/NZS 3260	Australia
TS001	Australia
NOM 016	Mexico
NOM 019	Mexico

Network standards

Table 3: Network Standards

Standard	Country/Region
CSO3	Canada
FCC Part 68	USA
TBR4	Europe
TBR4 Appendix 1 for Layer 3 Testing	New Zealand
TBR12	Europe
TBR13	Europe
TBR21	Europe
TS002	Australia
TS014	Australia
TS016	Australia
TS038	Australia
JATE	Japan
NOM151	Mexico
NOM152	Mexico
HKTA 2011	Hong Kong
HKTA 2013	Hong Kong
HKTA 2015	Hong Kong
HKTA 2017	Hong Kong
HKTA 2018	Hong Kong
HKTA 2023	Hong Kong
HKTA 2028	Hong Kong

EMC standards

Table 4: EMC Standards

Standard	Country/Region
FCC PART 15, Class A	USA
ICES 003, Class A	Canada
AS/NZS 3548, Class B	Australia, New Zealand
EN55022, Class B	Europe
EN55024	Europe
EN61000-3-2	Europe
EN61000-3-3	Europe
VCCI, Class B	Japan
Plug and Power Specifications	Argentina

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Let us know what you like or don't like about this book. Although we can't respond personally to all your feedback, we promise we will read each response we receive.

Write to us at:

Avaya Inc.
Product Documentation Group
Room B3-H13
1300 W. 120th Ave.
Westminster, CO 80234 USA

Fax to:

303-538-1741

Send email to:

document@avaya.com

Chapter 1: Networking overview

This chapter provides background information to help you understand and use the information in this book. Telephony delivered over digital networks capitalizes on the flexibility of technology itself, and can be implemented in a variety of ways. Users might find that they need to reference only a portion of the information in this book; others might need most of its information before understanding how to tailor a telephony network to suit their needs.

What is a network

An Avaya Communication Manager *network* can contain multiple interconnected media servers and all of the equipment, including data networking devices, under those media servers' control. Such equipment may be geographically dispersed among a variety of sites, and the equipment at each site may be segregated into distinct logical groupings, referred to as *network regions*. A single media server system is comprised of one or more *network regions*. Each *network region* is a logical grouping of endpoints, including stations, trunks, and media gateways. In cases where one media server is insufficient for controlling all of the equipment, multiple systems can be networked together. So, a *network region* is a component of a *site*, which is a component of a *system*, which is a component of a *network*.

About "network" terminology

For the purposes of this book and to clarify what we mean by the word, consider these uses of the word "network":

- Businesses often have a "customer network," meaning a Local Area Network (LAN) or a Wide Area Network (WAN), over which they distribute E-mail, data files, run applications, access the Internet, and send and receive fax and modem calls.

  We use *non-dedicated* to describe this type of network and the traffic that it bears. This means that the network is a heterogeneous mix of data types.

- When a non-dedicated network carries digitized voice signals along with other mixed-data types, we call this a *converged* network, because it is a confluence of voice and non-voice data.

- Network segments that exclusively carry telephony traffic are *dedicated*, since they carry only telephony-related information.

  The section What's in a digital phone call describes the types of data that are exchanged through dedicated networks.

- When a digital network carries telephony and non-telephony data in a packet-switched (TCP/IP), instead of a circuit-switched (TDM) environment, we call this an *IP network*.

What's in a digital phone call

A digital phone call consists of voice (bearer) data and call-signaling messages. Some transmission protocols require sending signaling data over a separate network, virtual path, or "channel," from the voice data. The following list describes the data that are transmitted between switches during a phone call:

- Voice (bearer) data — digitized voice signals
- Call-signaling data — control messages
  - Set up the call connection
  - Maintain the connection during the call
  - Tear down the connection when the call is finished
- Distributed Communications System (DCS) signaling data — an Avaya DEFINITY® Server proprietary signaling protocol.

  Distributed Communications System (DCS) allows you to configure 2 or more switches as if they were a single, large switch. DCS provides attendant and voice-terminal features between these switch locations. DCS simplifies dialing procedures and allows transparent use of some of the Communication Manager features. (Feature transparency means that features are available to all users on DCS regardless of the switch location.)

  Note:

  DCS is different from call-signaling data. See Distributed Communications System on page 163 for more information.

About network regions

A network region is a group of IP endpoints that share common characteristics and resources. Every IP endpoint on an Avaya Communication Manager system belongs to a network region. By default all IP endpoints are in network region 1, and if left as such they would all share the same characteristics defined by network region 1, and use the same resources. But in many cases this is not sufficient, and multiple network regions should be configured.

The most common of these cases are:

- One group of endpoints requires a different codec set than another group.

  This could be based on requirements related to bandwidth or encryption.

- Calls between separate groups of endpoints require a different codec set than calls within a single group of endpoints, again based on requirements related to bandwidth or encryption.
- Specific C-LAN or MedPro or other resources must be accessible to only a specific group of endpoints.
- One group of endpoints requires a different UDP port range or QoS parameters than another group.
- One group of endpoints reports to a different VoIP Monitoring Manager server than another group.

Somewhat related to network regions is the concept of locations. The *location* parameter is used to identify distinct geographic locations, primarily for call routing purposes. In other words, the location parameter is used primarily to insure that calls access the proper trunks, based on the origin and destination of each call.

Establishing inter-switch trunk connections

Avaya equipment is configured in different ways for various reasons. Connected switches enable people within an enterprise to communicate easily with one another, regardless of their physical location or the particular communications server they use. Inter-switch connections also provide shared communications resources such as messaging and Call Center services.

Switches communicate with each other over trunk connections. There many types of trunks that provide different sets of services. Commonly-used trunk types are:

- Central Office (CO) trunks that provide connections to the public telephone network through a central office.
- H.323 trunks that transmit voice and fax data over the Internet to other systems with H.323 trunk capability.

  H.323 trunks that support DCS+ and QSIG signalling.
- Tie trunks that provide connections between switches in a private network.

These and other common trunk types are described in the *Administrator's Guide for Avaya Communication Manager, 555-233-506*.

Interconnecting port networks

Multi-Connect systems with more than four port networks (PNs) may use a center stage switch (CSS) to interconnect the PNs. It is now possible to separate the CSS geographically in two types of configurations.

For systems with more than 16 PNs, requiring more than a single switch node carrier (SNC), the first switch node (1A) is at location 1 and the second switch node (2A) is at location 2. For duplicated systems, the duplicate switch nodes are similarly separated. These duplicate switch nodes (1A to 2A) and (1B to 2B) are linked together by fiber connections between locations.

In the second configuration type, the duplicated switch nodes are separated. The switch nodes 1A and 2A are at location one, and the switch nodes 1B and 2B are at location 2. At each location the switch nodes are linked together by fiber connections.

For more information, see: http://support.avaya.com/elmodocs2/s8700/PDFs/ServSeparate-EqWithCSSseparationFINAL.pdf.

Networking branch offices

For Avaya Communication Manager environments, The MultiVOIP™ voice over IP gateways provide distributed networking capabilities to small branch offices of large corporations. MultiVOIP extends the call features of a centralized Avaya Media Server and provides local office survivability to branch offices of up to 15 users using analog or IP phones.

For more information, see: http://www.multitech.com/partners/avaya/Avaya%20DevConnDatasheet.pdf.

Enabling spanning tree (STP)

Spanning Tree (STP) is a loop avoidance protocol. If you don't have loops in your network, you don't need STP. The "safe" option is to always leave STP enabled. Failure to do so on a network with a loop (or a network where someone inadvertently plugs the wrong cable into the wrong ports) will lead to a complete cessation of all traffic.

STP is slow, however. It is slow to converge after a network failure, and slow to allow a new port into the network (~50 sec by default). In early generations of phones, this delay caused phone DHCP requests to time out, and the phones didn't recover gracefully, so there was a recommendation to use "port fast" (Cisco) or "fast start" (some Cajuns). Because Cajun P330s didn't support those features, the recommendation was modified to *disable* STP on phone ports on those switches. This soon came to be understood as disable STP everywhere, which was not necessary.

Since that time, Rapid Spanning Tree emerged, which converges faster than the earlier STP, and enables new ports much faster (sub-second) than the older protocol. **Rapid Spanning Tree** works with all Avaya equipment, and can be *recommended*.

Network quality management

A successful Voice over Internet Protocol (VoIP) implementation involves quality of service (QoS) management that is impacted by three major factors:

- *Delay:* Significant end-to-end delay may result in echo and talker overlap.

  Echo becomes a problem when one-way network delay is more than 50 milliseconds. VoIP systems must implement some means of echo cancellation. If the round-trip delay is greater than 250 milliseconds (ms), talker overlap, or one caller “stepping on” the other talker's speech, is likely. For adequate quality of service, we recommend that the total network delay should be less than 50ms. See Packet delay and loss on page 121 for more information.

- *Packet Loss:* Under peak network loads and periods of congestion, voice data packets may be dropped.

  Because voice transmission is highly time-sensitive, normal TCP-based re-transmission schemes are not suitable. Methods to compensate for packet loss include interpolation of speech by re-playing the last packet and sending of redundant information. The maximum packet loss between network endpoints should not exceed 0.2%. See Packet delay and loss on page 121 for more information.

- *Jitter (Delay Variability):* Jitter results when data packets arrive at their destination at irregular intervals as a result of variable transmission delay over the network.

  To remove jitter, the VoIP engine must collect and hold data packets in a buffer long enough for the slowest packet to arrive and be played in sequence. A jitter buffer, however, adds to delay. Jitter of less than 20ms. between endpoints is normally required. See Packet delay and loss on page 121 for more information.

Sending and receiving IP packets

Prior to transmission over an IP network a voice signal is converted from analog to digital form, usually at a rate of 8,000 samples per second. Then the digital bit stream is sampled or compressed through A-law, mu-law, or bit-rate companding methods, and finally grouped into packets for transmission. To use network bandwidth efficiently, a silence suppression algorithm that detects when there are periods of silence does not transmit packets in those brief spaces.

When the packets are received, several processes occur:

- Packets are put in proper order and converted back to an analog voice signal
- Jitter is removed
- The effects of packet loss are mitigated through various algorithms
- Silence suppression is eliminated by adding artificial samples, often in the form of comfort noise, a random, low-level signal that gives the impression that the connection is still alive during periods of silence.
- Echo-cancellation eliminates acoustic or electronic network reflection effects.

About VoIP-transmission hardware

For more detailed descriptions and administration procedures for the equipment listed below, see About VoIP hardware on page 39.

TN799 (C-LAN)

The C-LAN circuit pack provides the data link interface between the switch processor and the transmission facilities. C-LAN prepares the signaling information for TCP/IP transmission over one of two pathways:

- Ethernet connection — the signaling data is sent out on a 10/100Base-T network, which is connected directly to the C-LAN Ethernet port.
- PPP connection — C-LAN inserts the signaling data on the TDM bus for subsequent inclusion (through the switching fabric) in the same DS1 bit stream as the voice transmissions.

TN802B (IP-Interface)

The IP Interface circuit pack (TN802B) enables two switches to transmit voice between them over an IP network. The TN802B normally operates in the MedPro mode, which supports H.323-compliant endpoints or applications.

TN2302AP (IP Media Processor)

The TN2302AP transmits voice and fax data (non-DCS signaling) over IP connections. It also improves quality of service through its dynamic jitter buffers, echo cancellation, and silence suppression, making it suitable for H.323 multimedia applications and other H.323-compliant endpoints. This circuit pack also performs DTMF detection and conferencing.

The TN2302AP IP Media Processor can work in the same server with the TN802B IP Interface Assembly. The software chooses media processing resources for an IP endpoint from the TN2302AP over the TN802B when both types of media processing are available on the system.

TN2312 (IP Server Interface)

The TN2312 IP Server Interface circuit pack (IPSI) provides an interface between the Avaya Media Servers and up to 5 port networks. The IPSI connects to the S8700 Media Server through Ethernet.

G700/G350 Media Gateway VoIP processors

The VoIP processor on the G700 Media Gateway motherboard performs IP/UDP/RTP processing, echo cancellation, G.711 A-/µ-law, G.729 and G723.1 encode/decode, FAX relay, silence suppression, jitter buffer management, and packet-loss concealment. The VoIP Engine supports 64 channels. If more than 64 channels are needed, an MM760 VoIP Media Module is required.

The G350 Media Gateway supports the G.711 codec for up to 32 concurrent calls and the G.729 codec for up to 16 concurrent calls.

MM760 VoIP Media Module

The MM760 VoIP Media Module is a duplicate of the VoIP processor in the G700 motherboard. The capacity of the MM760 is 64 G.711 TDM/IP simultaneous calls, or 32 compression codec, G.729 or G.723, TDM/IP simultaneous calls. These call types can be mixed on the same resource. In other words, the simultaneous call capacity of the resource is 64 G.711 Equivalent Calls.

About call connections

Table 5: Call connections and data transmissions summarizes the different types of call connections and how the voice and signaling data are transmitted between switches.

Table 5: Call connections and data transmissions

Connection Type	Tie Trunk		LAN or WAN
	Voice & Call-Signaling	DCS Signaling	Voice
DCS+ over ISDN	T1/E1 facilities, ISDN-PRI B-Channel	TSCs[1] on the ISDN-PRI D-Channel	
QSIG[2] over ISDN	T1/E1 facilities, ISDN-PRI B-Channel	CISCs[3] on the ISDN-PRI D-Channel	
X.25	T1/E1 facilities using ISDN-PRI or DS1 B-Channel or analog trunk	Packet PVC[4]	
C-LAN PPP			
C-LAN Ethernet		TCP datagrams through IP circuit packs (see About VoIP-transmission hardware)	
IP Interface			RTP[5] Packet IP media processor

[1] TSC — Temporary Signaling Connection.

[2] QSIG — Signaling standard used for networks containing dissimilar switches.

[3] CISC — Call-independent Signaling Connection provides a temporary signaling path through ISDN switches for exchanging supplementary service information.

[4] PVC — Permanent virtual circuit provides for a permanent switched call (PSC) to be established and maintained between two data endpoints that should always be connected.

[5] RTP — Real-time Transport Protocol provides end-to-end network transport functions for real-time audio or video over multicast or unicast network services.

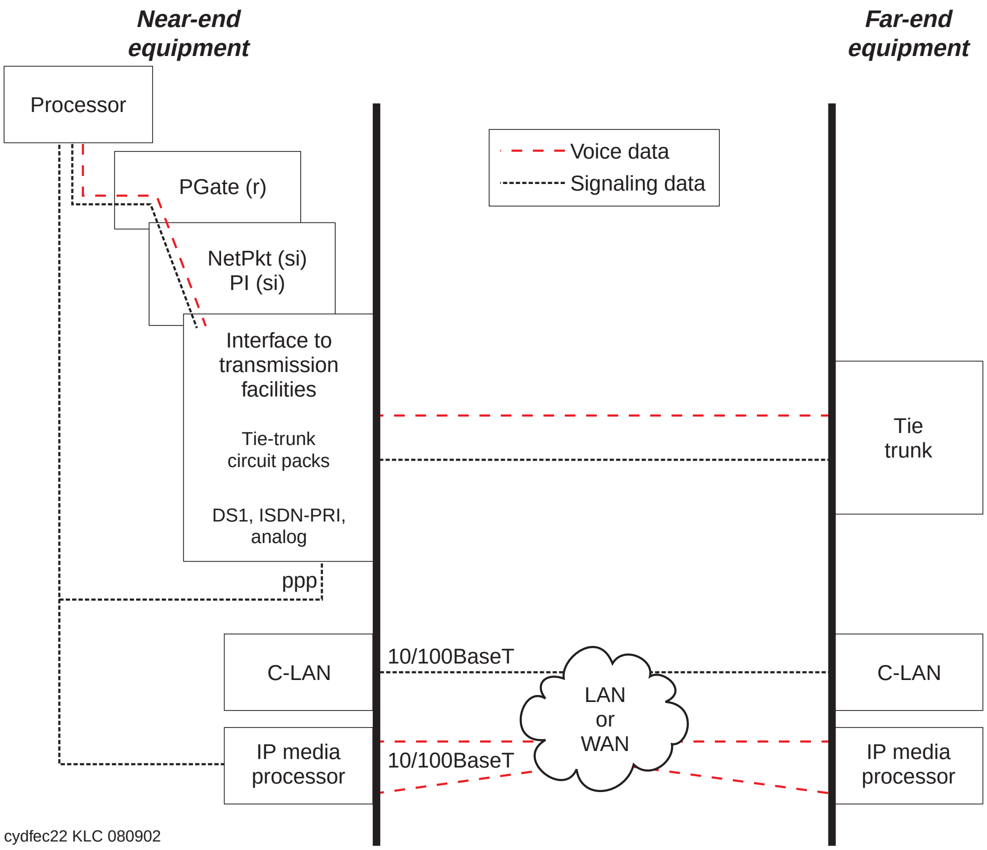
For DCS+, X.25, and PPP connections, the signaling and voice data are sent together over tie-trunk facilities as TDM-multiplexed frames. DCS signaling data are sent as packets over a permanent virtual circuit (PVC) on tie-trunk facilities.

For VoIP/Ethernet connections, the signaling and voice data can be sent either:

- Signaling over IP; voice bearer over circuit-switched
- Both over IP

The DCS signaling data are sent as transmission control protocol (TCP) datagrams over an IP network through any of the circuit packs listed in the About VoIP-transmission hardware section.

Figure 1: Switch connection components



Connection types

This section gives an overview of the types of network connections that Communication Manager can establish.Table 6: Types of connections on page 32 lists the types of connections possible with each model and adjunct.

For a complete list of the types of connections possible with each model, go to the *Hardware Guide for Avaya Communication Manager, 555-245-207*, in the subsection titled "Circuit packs and power supplies", and do a keyword search through the PDF file for whatever communication protocol in which you are interested. In addition, you will get a list of the circuit packs that carry the signaling for that protocol.

Table 6: Types of connections ***1 of 3***

Server Type	Connection Type	Endpoint or Service
Avaya DEFINITY Server csi	Ethernet	DCS, CMS, Intuity AUDIX, IP Telephone, IP Softphone
	Synchronous PPP	DCS
	ISDN-PRI	DCS+, QSIG
	H.323 Trunk	DCS+, QSIG
Avaya DEFINITY Server si	Ethernet	DCS, CMS, Intuity AUDIX, IP Telephone, IP Softphone
	Synchronous PPP	DCS
	ISDN-PRI	DCS+, QSIG
	X.25	DCS, CMS, Intuity AUDIX, DEFINITY AUDIX
	H.323 Trunk	DCS+, QSIG
Avaya DEFINITY Server r	Ethernet	DCS, CMS, Intuity AUDIX, IP Telephone, IP Softphone
	Synchronous PPP	DCS, CMS, Intuity AUDIX
	ISDN-PRI	DCS+, QSIG
	X.25	DCS, CMS, Intuity AUDIX, DEFINITY AUDIX
	H.323 Trunk	DCS+, QSIG
Avaya S8300 Media Server	Ethernet	DCS+, Intuity AUDIX, Embedded AUDIX, IP Telephone
	Mode Code	Intuity AUDIX
	ISDN-PRI	DCS+, QSIG
	H.323 Trunk	DCS+, QSIG

1 of 3

Table 6: Types of connections ***2 of 3***

Server Type	Connection Type	Endpoint or Service
Avaya S8500 Media Server	Ethernet	DCS+, Intuity AUDIX, Embedded AUDIX, IP Telephone
	MAPD	Intuity AUDIX
	Mode Code	Intuity AUDIX
	ISDN-PRI	DCS+, QSIG
	H.323 Trunk	DCS+, QSIG
Avaya S8700 Media Server	Ethernet	DCS+, Intuity AUDIX, Embedded AUDIX, IP Telephone
	MAPD	Intuity AUDIX
	Mode Code	Intuity AUDIX
	ISDN-PRI	DCS+, QSIG
	H.323 Trunk	DCS+, QSIG
Avaya S8710 Media Server	Ethernet	DCS+, Intuity AUDIX, Embedded AUDIX, IP Telephone
	MAPD	Intuity AUDIX
	Mode Code	Intuity AUDIX
	ISDN-PRI	DCS+, QSIG
	H.323 Trunk	DCS+, QSIG
Avaya G700 Media Gateway	Ethernet	DCS+, Intuity AUDIX, Co-resident AUDIX, IP Telephone
	ISDN-PRI	MM710 E1/T1
	PPP	X330 WAN E1/T1
	H.323 Trunk	MM760 VoIP

2 of 3

Table 6: Types of connections ***3 of 3***

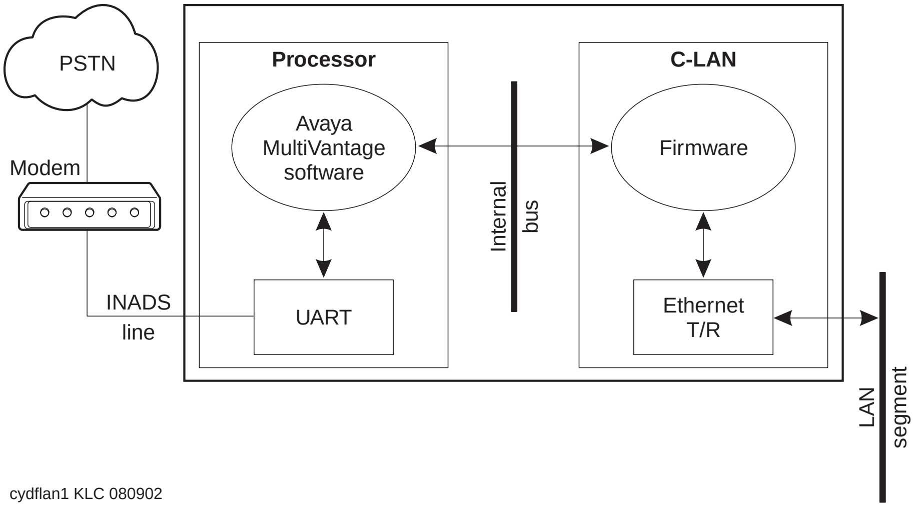
Server Type	Connection Type	Endpoint or Service
Avaya G350 Media Gateway	Ethernet	PoE, DCS+, Intuity AUDIX, Co-resident AUDIX, IP Telephone
	ISDN-PRI	MM710 E1/T1 Trunk
	PPP	E1/T1 Trunk
Avaya C360 Ethernet Switch	Ethernet	DCS+, Intuity AUDIX, IP Telephone

3 of 3

Providing LAN security

Some customers are concerned that a user could access the switch using the INADS line, gain access to C-LAN, and then access to the customer's LAN. The Avaya architecture prevents access to the customer's LAN as depicted in Figure 2: Security-related system architecture on page 34, which shows a high-level switch schematic with a TN799 (C-LAN) circuit pack.

Figure 2: Security-related system architecture



Logins through the INADS line terminate in software; software communicates with firmware over an internal bus through a limited message set. There are two main reasons why a user cannot access a customer's LAN through the INADS line:

- A user logging into software cannot obtain direct access to the C-LAN firmware.

  The user can only enter SAT commands that request C-LAN information or to configure C-LAN connections.
- The C-LAN application TFTP is currently disabled and cannot be enabled by Avaya Communication Manager.

  TELNET only interconnects C-LAN Ethernet clients to the system management application on the switch. FTP exists only as a server, is used only for firmware downloads, and it cannot connect to the client network.

For more information about LAN security, see

- *Security Topics for the Avaya S8700 Media Server Configurations*
- *Security for the Avaya 8700 Media Server*

Chapter 2: Administering converged networks

This section provides information for administering converged network components.

- About Voice over IP converged networks
- Providing a network assessment
- About VoIP hardware
- Adding a G600/G650 Media Gateway to an S8500 or S8700-series Media Server
- Adding a G350, or G700 Media Gateway to an S8300 Media Server
- Administering H.323 trunks
- Administering IP Softphones
- Installing and administering Avaya IP telephones
- About hairpinning and shuffling

About Voice over IP converged networks

Until recently, voice, video, and data were delivered over dedicated, single-purpose networks. A converged network brings all forms of communication together: voice, data, and video. Avaya's products use Voice over IP (VoIP) technology as their means of connection. VoIP technology provides a cost-effective and flexible way of building enterprise communications systems through a converged network.

Some of the flexible elements include:

- Separation of call control and switching functions (see the *Separation of Bearer and Signaling Job Aid*, *555-245-770,* on the library CD, 555-233-825)
- Different techniques for handling data, voice, and fax
- Communications standards and protocols for different network segments
- Constant and seamless reformatting of data for differing media streams

Digital data and voice communications superimposed in a converged network compete for the network bandwidth, or the total information throughput that the network can deliver. Data traffic tends to require significant network bandwidth for short periods of time, while voice traffic demands a steady, relatively constant transmission path. Data traffic can tolerate delays, while voice transmission degrades, if delayed. Data networks handle data flow effectively, but when digitized voice signals are added to the mix, networks must be managed differently to ensure constant, real-time transmission needed by voice.

Providing a network assessment

Even if your network appears to perform acceptably, adding VoIP taxes network resources and performance, because VoIP requires dedicated bandwidth and is more sensitive to network problems than data applications alone. Many customer IP infrastructures appear to be stable and perform at acceptable levels but have performance and stability issues that create problems for Avaya VoIP Solutions. While the customer network appears to be ready for full-duplex VoIP, Avaya cannot assure performance and quality without a network assessment.

The network assessment services for Avaya VoIP consist of 2 phases:

- Customer Infrastructure Readiness Survey (CIRS) — is a high-level LAN/WAN infrastructure evaluation that determines the suitability of an existing network for VoIP.
- Network Analysis Network Optimization (NANO) — is typically the second phase in the Network Assessment for IP Telephony solutions.

  The NANO takes information gathered from the CIRS, performs problem diagnosis, and provides functional requirements for the network to implement Avaya VoIP.

Avaya Network Consulting Services (NCS) supports a portfolio of consulting and engineering offers to help plan and design:

- IP Telephony
- Data Networking Services
- Network Security Services.

How to contact the NCS

- On the Web -- http://ncs.avaya.com/
- E-Mail: ncs@avaya.com
- Phone: +1 866-832-0925

For full details on these services, see the chapter on "Network assessment offer" in *Avaya Application Solutions: IP Telephony Deployment Guide*, 555-245-600

About VoIP hardware

This section contains descriptions and administration for the following circuit packs and media modules:

- TN464GP Universal DS1 circuit pack and MM710 T1/E1Media Module
- TN799DP Control LAN
- TN802B MAPD (IP Interface Assembly)
- TN2302AP IP Media Processor
- TN2312BP IP Server Interface (IPSI)
- MM760 VoIP Media Module

TN464GP Universal DS1 circuit pack and MM710 T1/E1Media Module

The TN464GP circuit pack and the MM710 Media Module (version 3 and later) have the same functionality as other DS1 circuit packs, with the addition of echo cancellation circuitry. The TN464GP and MM710 offer echo cancellation tail lengths of up to 96 milliseconds (ms). The TN574, TN2313, and TN2464 DS1 circuit packs do not support echo cancellation.

The TN464GP and MM710 are intended for users who encounter echo over circuits connected to the Direct Distance Dialing (DDD) network. Echo is most likely to occur when Avaya Communication Manager is configured for ATM, IP, and wideband. In addition, echo can occur on system interfaces to local service providers that do not routinely install echo cancellation equipment in all their circuits.

Echo cancellation is a software right-to-use feature that supports voice channels, and is not intended for data. When a data call is received, these circuit packs detect a modem tone and turn off echo cancellation for the duration of the data call. In addition, echo cancellation is selectable per channel, even though it is administered on a trunk group basis.

Working with echo cancellation

You can determine whether echo cancellation is enabled for TN464GP circuit packs and MM710 T1/E1 Media Modules using the **system-parameters customer-options screen**.

To determine if echo cancellation is enabled for TN464GP circuit packs and MM710 T1/E1 Media Modules

1. Type **`display system-parameters customer-options`**.

   Note:

   > The **Customer Options** screen is display-only. The License File controls the system software release, the Offer Category, features, and capacities. The *init* login cannot change the customer options, offer options, or special applications screens, unless a feature is enabled but not turned on in the License File.

2. Find and review the following fields.

   The fields may appear on different pages of the screen.

Field	Conditions/Comments
Maximum Number of DS1 Boards with Echo Cancellation	Specifies the number of DS1 boards that have echo cancellation turned on.
DS1 Echo Cancellation	If **y**, echo cancellation is enabled.

3. Exit the screen.

Administering echo cancellation on the DS1 circuit pack or MM710 media module

The **DS1 Circuit Pack** screen for the TN464GP circuit pack and MM710 media module has fields to support echo cancellation: **Echo Cancellation**, **EC Direction**, and **EC Configuration**. The **Echo Cancellation** field appears when the Echo Cancellation feature is activated on the **System-Parameters Customer Options** screen. The **EC Direction** and **EC Configuration** fields appear when the **DS1 Echo Cancellation** field is enabled.

- **EC Direction** determines the direction from which echo will be eliminated, ether inward or outward.
- **EC Configuration** is the set of parameters used when cancelling echo.

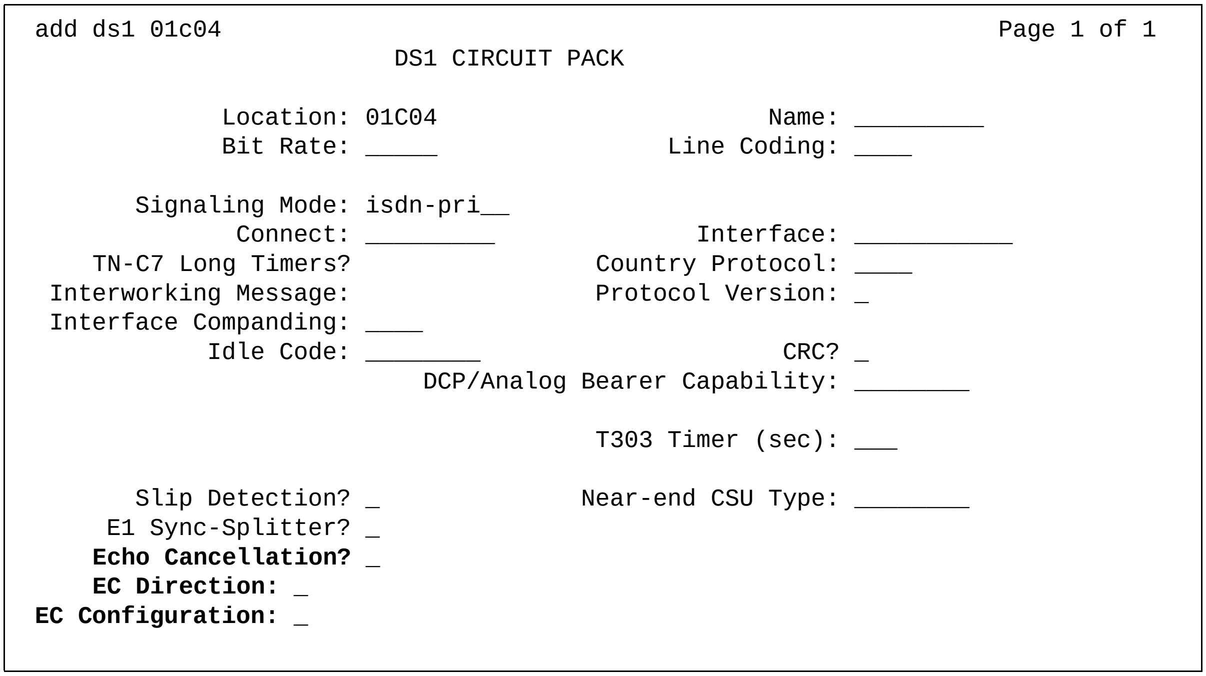
  This information is stored in firmware on the UDS1 circuit pack.

To administer the DS1 circuit pack and MM710 media module

1. Type **`add ds1 <port>`** and press **Enter** to open the **DS1 Circuit Pack** screen,

   where ***`<port>`*** is the location of the DS1 circuit pack, or the MM710 media module.

DS1 Circuit Pack screen



```
add ds1 01c04                                                       Page 1 of 1
                        DS1 CIRCUIT PACK

             Location: 01C04                           Name: _________
             Bit Rate: _____                    Line Coding: ____

       Signaling Mode: isdn-pri__
              Connect: _________                  Interface: ___________
    TN-C7 Long Timers?                     Country Protocol: ____
  Interworking Message:                    Protocol Version: _
  Interface Companding: ____
            Idle Code: ________                        CRC? _
                              DCP/Analog Bearer Capability: ________

                                           T303 Timer (sec): ___

       Slip Detection? _                  Near-end CSU Type: ________
     E1 Sync-Splitter? _
    Echo Cancellation? _
    EC Direction: _
EC Configuration: _
```

2. On the **DS1 Circuit Pack** screen, complete the following fields:

Field	**Conditions/Comments**
Echo Cancellation	Enter **y** to enable echo cancellation on the Universal DS-1 circuit pack.

Field	Conditions/Comments
EC Direction	Indicates the direction of the echo that is being cancelled. Enter **inward** or **outward**. ● The **inward** setting cancels echo energy coming back into the switch — energy from an outgoing call is reflected from an external reflection point (party "inside" the switch hears the echo). ● The **outward** setting cancels echo energy going outside the switch — energy from an incoming call is reflected from an internal reflection point (party "outside" the switch hears the echo).
EC Configuration	Indicates the set of echo cancellation defaults to administer. Appears when the Echo Cancellation field is set to **y**. Enter digits between **1-15**. ● Enter **1** or **5-15** to provide most rapid adaptation in detecting and correcting echo at the beginning of a call, regardless of the loudness of the talker’s voice. For very loud talkers and severe echo, the far-end talker’s speech is heard as clipped when both parties talk at the same time. ● Enter **2** for slightly slower adaptation to echo, use if speech is often clipped when both parties talk at the same time. ● Enter **3** for slightly slower adaptation to echo, may result in a 2 or 3 second fade on strong echo for quiet talkers. Completely removes speech clipping. ● Enter **4** in cases of extreme echo, excessive clipping or breakup of speech. May result in slight echo or background noise. **Note:** For the MM710, the values **1** and **4** are reversed. That is, **1** for the MM710 is the same as **4** for the TN464GP, and **4** for the MM710 is the same as **1** for the TN464GP

Administering echo cancellation on trunks

Echo cancellation is turned on or off on a per trunk-group basis using the **`change trunk-group`** command. If the trunk group field, **DS1 Echo Cancellation** is y, echo cancellation is applied to every TN464GP trunk member in that trunk group. The echo cancellation parameters used for a given trunk member are determined by the **EC Configuration** number administered on the **DS1 Circuit Pack** screen for that specific trunk's board.

Echo cancellation applies to voice channels and supports echo cancellation on the following trunk group types:

- CO
- TIE
- ISDN-PRI
- FX
- WATS
- DID
- DIOD
- DMI-BOS
- Tandem
- Access
- APLT

Administration of echo cancellation on a trunk group is done on the **TRUNK FEATURES** screen.

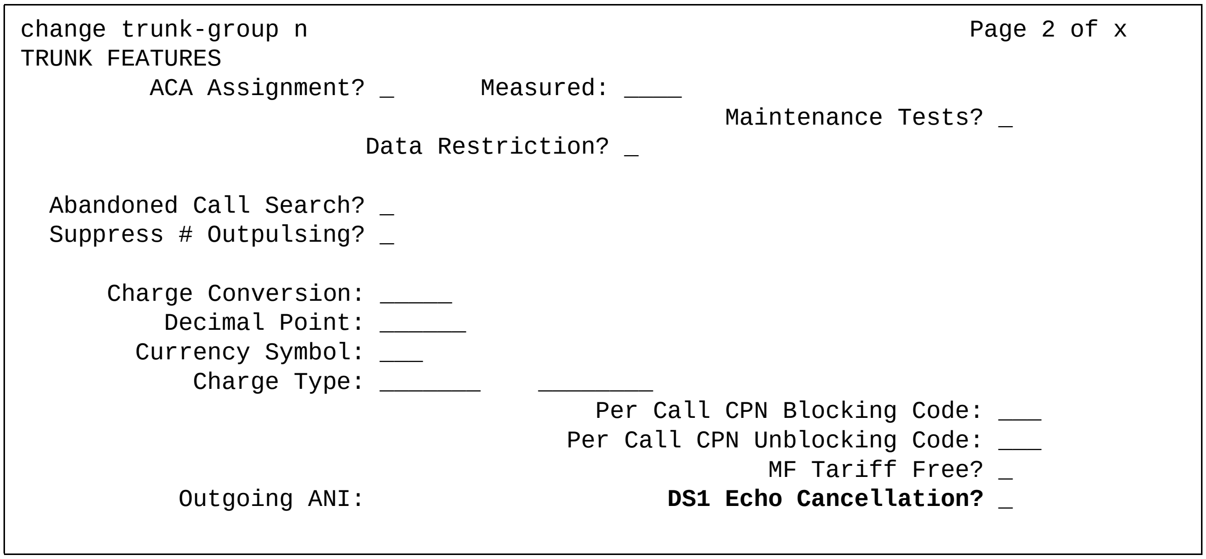
To administer a trunk group for echo cancellation

1. Type **`change trunk-group`** ***n***

   where ***n*** is the trunk group number.

2. Go to page 2.

Trunk Features screen



```
change trunk-group n                                              Page 2 of x
TRUNK FEATURES
         ACA Assignment? _      Measured: ____
                                                 Maintenance Tests? _
                        Data Restriction? _

  Abandoned Call Search? _
  Suppress # Outpulsing? _

      Charge Conversion: _____
          Decimal Point: ______
        Currency Symbol: ___
            Charge Type: _______    ________
                                        Per Call CPN Blocking Code: ___
                                      Per Call CPN Unblocking Code: ___
                                                    MF Tariff Free? _
           Outgoing ANI:                     DS1 Echo Cancellation? _
```

3. Move to the following field

Field	Conditions/Comments
DS1 Echo Cancellation	Enter **y** to enable echo cancellation on a per trunk group basis.

4. Save the changes.

   Changes to the **DS1 Echo Cancellation** field do not take effect until one of the following occurs:

   - Port is busied-out/released.
   - Trunk group is busied-out/released.
   - SAT command **`test trunk group`** is performed.
   - Periodic maintenance runs.

Administering echo cancellation per channel

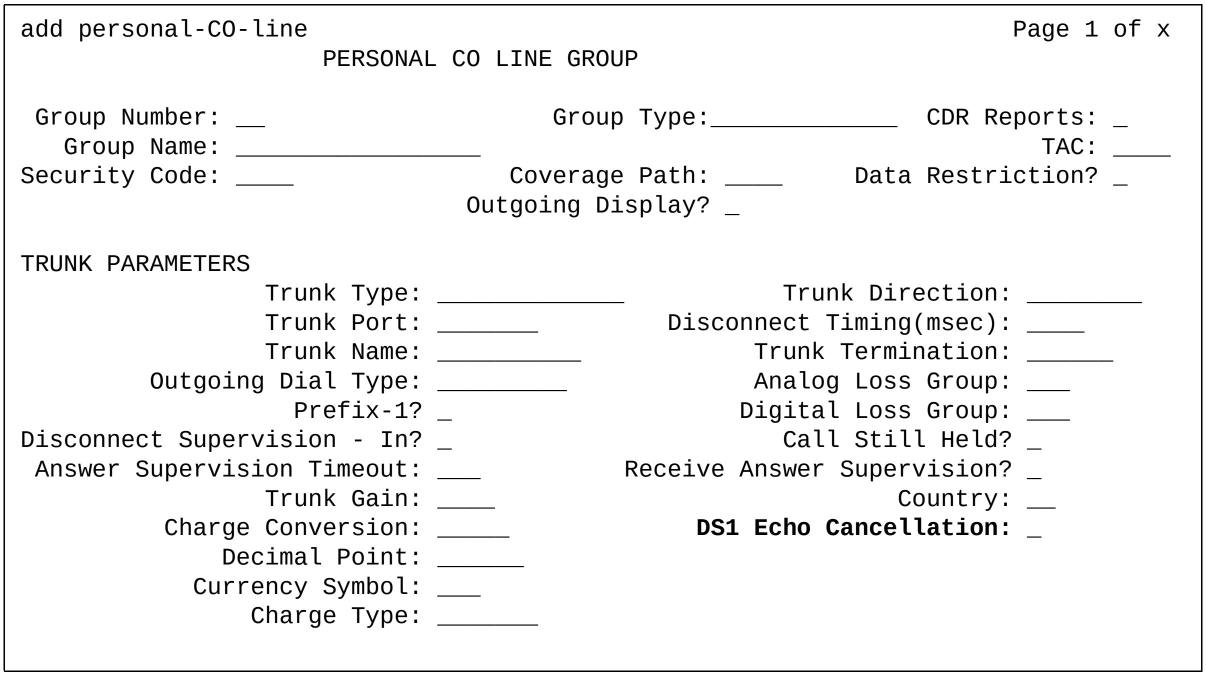
Echo cancellation on the TN464GP or MM710 media module is selectable per channel, even though it is administrable on a per trunk-group basis. For example, if all but two ports on a TN464GP or MM710 need to have echo cancellation applied, those two ports must be put in a trunk group where the **DS1 Echo Cancellation** field is **n**. The remaining ports are in a trunk group(s) where the **DS1 Echo Cancellation** field is **y**.

A user can cancel echo coming from the network (far-end echo) or coming from the switch (near-end echo).

To administer echo cancellation per channel

1. Type **`add personal-co-line`** and press **Enter** to open the **Personal CO Line Group** screen.

Personal CO Line Group screen



```
add personal-CO-line                                             Page 1 of x
                     PERSONAL CO LINE GROUP

 Group Number: __                    Group Type:_____________  CDR Reports: _
   Group Name: _________________                                       TAC: ____
Security Code: ____               Coverage Path: ____     Data Restriction? _
                               Outgoing Display? _

TRUNK PARAMETERS
                 Trunk Type: _____________           Trunk Direction: ________
                 Trunk Port: _______         Disconnect Timing(msec): ____
                 Trunk Name: __________            Trunk Termination: ______
         Outgoing Dial Type: _________             Analog Loss Group: ___
                   Prefix-1? _                    Digital Loss Group: ___
Disconnect Supervision - In? _                       Call Still Held? _
 Answer Supervision Timeout: ___          Receive Answer Supervision? _
                 Trunk Gain: ____                            Country: __
          Charge Conversion: _____             DS1 Echo Cancellation: _
              Decimal Point: ______
            Currency Symbol: ___
                Charge Type: _______
```

2. Complete the following field:

Field	Conditions/Comments
DS1 Echo Cancellation	Enter **y** to enable echo cancellation on a Personal CO Line.

3. Submit the screen.

TN799DP Control LAN

Systems in a private network are interconnected by both tie trunks (for voice communications) and data links (for control and transparent feature information). Various DS1, IP, and analog trunk circuit packs provide the voice-communications interface. For TCP/IP connectivity, the data-link interface is provided by a TN799DP Control LAN (C-LAN) circuit pack.

The C-LAN handles the data-link signaling information in one of two configurations: Ethernet, or point-to-point (PPP). The C-LAN circuit pack has one 10/100baseT ethernet connection and up to 16 DS0 physical interfaces for PPP connections. C-LAN also extends ISDN capabilities to csi models by providing packet-bus access.

- In the Ethernet configuration, the C-LAN passes the signaling information over a separate TCP/IP network, usually by means of a hub or Ethernet switch.

  Avaya recommends an Ethernet switch for optimal performance. For this configuration, install the C-LAN circuit pack and connect the appropriate pins of the C-LAN I/O field to the hub or Ethernet switch.

- In the PPP configuration, the C-LAN passes the data-link signaling to the DS1 for inclusion in the same DS1 bit stream as the DCS voice transmissions.

  For this configuration, install the C-LAN circuit pack; no other connections are needed. The appropriate DS1 circuit packs must be installed, if they are not already present.

Physical addressing for the C-LAN board

The Address Resolution Protocol (ARP) on the C-LAN circuit pack relates the 32-bit IP address configured in software to the 48-bit MAC address of the C-LAN circuit pack. The MAC address is burned into the board at the factory. The C-LAN board has an ARP table that contains the IP addresses associated with each hardware address. This table is used to route messages across the network. Each C-LAN board has one MAC address, one Ethernet address, and up to 16 PPP addresses.

IP addressing techniques for the C-LAN board

The C-LAN supports both Classless Inter-domain Routing and Variable-Length Subnet Masks. These addressing techniques provide greater flexibility in addressing and routing than class addressing alone.

Installing the TN799DP C-LAN

TCP/IP connections (Ethernet or PPP) require a TN799DP C-LAN circuit pack, unless your system has embedded Ethernet capabilities. Before you install the C-LAN circuit pack, be sure you understand the requirements of your LAN. Go to http://www1.avaya.com/enterprise/whitepapers/ip_networking.pdf and search for the white paper *Avaya IP Voice Quality Network Requirements (EF-LB1500)*.

The following steps describe installation for the TN799DP C-LAN.

To insert TN799DP C-LAN circuit packs

1. Determine the carrier/slot assignments of the circuit packs to be added.

   You can insert the C-LAN circuit pack into any port slot.

2. Insert the circuit packs into the slots specified in step 1.

   Note:

   You do not need to power down the cabinet to install a C-LAN circuit pack.

Administering the C-LAN bus bridge (Avaya DEFINITY Server csi only)

For the Avaya DEFINITY Server csi only, complete the following steps to administer the bus bridge for the C-LAN circuit pack. Only an Avaya representative using the *craft* or higher login can change the maintenance parameters.

Note:

> If there are 2 C-LAN circuit packs installed in this csi switch, administer the bus bridge for *only one* of them.

To administer the C-LAN bus bridge (Avaya DEFINITY Server csi only)

1. Type **`change system-parameters maintenance`**.
2. Move to the **Packet Intf2** field and enter **y**.
3. Enter the location of the C-LAN circuit pack in the **Bus Bridge** field

   (for example, **01a08** for cabinet 1, carrier A, and slot 8).
4. Enter the port bandwidths or use the defaults in the **Pt0**, **Pt1**, and **Pt2 Inter-Board Link Timeslots** fields.
5. Submit the screen.
6. Verify that the bus bridge LED is lit on the C-LAN circuit pack.

   This indicates that the packet bus is enabled.

Testing the packet bus and C-LAN circuit pack

In order to test the packet bus and the TN799DP C-LAN circuit pack, the cabinet needs an installed TN771D Maintenance/Test circuit pack.

To test the packet bus and C-LAN circuit pack

1. If there is no TN771D circuit pack in the cabinet, place one in a port slot.

   This is for testing purposes only, and you will remove the board when finished.
2. Enter **`test pkt port-network`** ***`1 long`***

   For more information about these tests, refer to the **test pkt command** section in *Maintenance Commands for Avaya Communication Manager 2.1, Media Gateways and Servers*, 03-300191.
3. If the TN771D circuit pack was already in the cabinet, leave it there.
4. If you added the TN771D circuit pack to the cabinet in order to test the TN799DP circuit pack, remove it from the cabinet.

Installing C-LAN cables to a hub or ethernet switch

In the Ethernet configuration, the C-LAN passes the signaling information over a separate TCP/IP network, usually by means of a hub or Ethernet switch. Connect the appropriate pins of the C-LAN I/O field to the hub or Ethernet switch.

To install C-LAN cables to a hub or ethernet switch

See Figure 3: Cable connection for C-LAN connectivity.

1. Connect the 259A connector to the backplane connector of the port slot containing the C-LAN circuit pack.
2. Connect the Category 5 UTP cable between the 259A connector and a hub or Ethernet switch.

   This connects port 17 on the C-LAN circuit pack to the LAN.

Figure 3: Cable connection for C-LAN connectivity

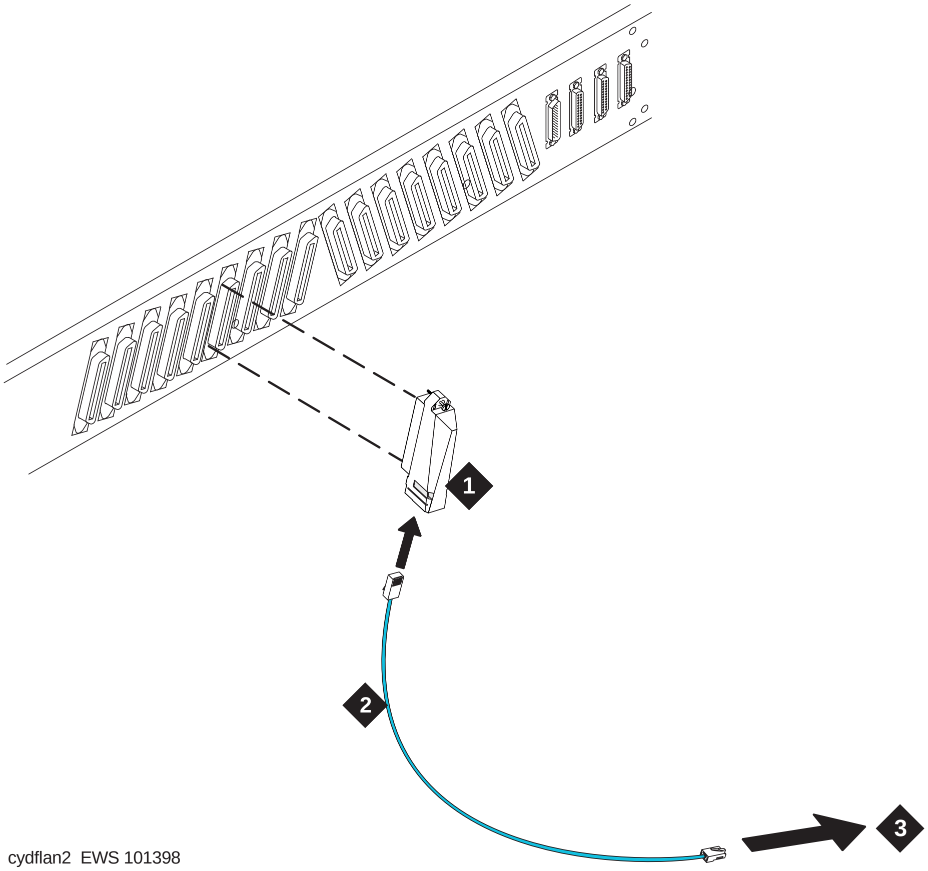


Figure notes:

1. **259A Connector**
2. **Category 5 UTP Cable (max length 100m)**
3. **Ethernet switch**

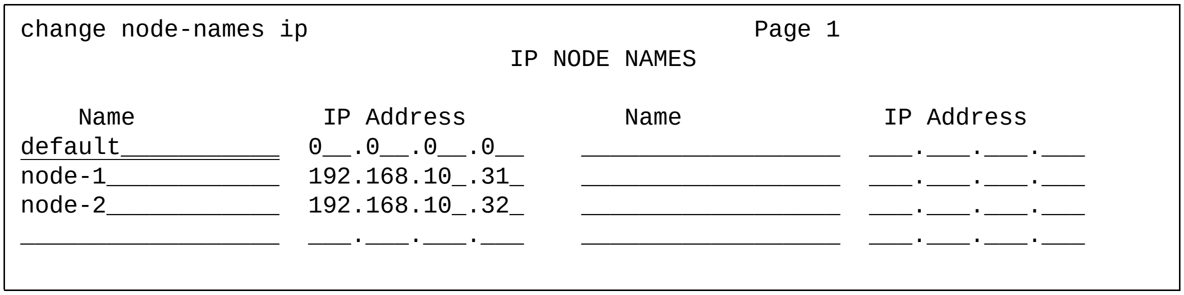
Assigning IP node names

You must assigns node names and IP addresses to each node in the network. Administer the **Node Names** screen on each switch in the network.

You should assign the node names and IP addresses logically and consistently across the entire network. These names and addresses should be assigned in the planning stages of the network and should be available from the customer system administrator or from an Avaya representative.

To assign IP node names

1. Type **change node-names ip** and press **Enter** to open the **IP Node Names** screen.



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

    Name             IP Address            Name             IP Address
default___________  0__.0__.0__.0__    __________________  ___.___.___.___
node-1____________  192.168.10_.31_    __________________  ___.___.___.___
node-2____________  192.168.10_.32_    __________________  ___.___.___.___
__________________  ___.___.___.___    __________________  ___.___.___.___
```

2. Enter values.

Field	Conditions/Comments
Name	Enter unique node names for each switch or adjunct that will connect to this switch through the CLAN board.
IP Address	The unique IP addresses of the nodes named in the previous field.

3. Submit the screen.

Defining a LAN default gateway

On LANs that connect to other networks or subnetworks, Avaya recommends that you define a default gateway. The default gateway node is a routing device that is connected to different (sub)networks. Any packets addressed to a different (sub)network, and for which no explicit IP route is defined, are sent to the default gateway node.

You use the **IP Interfaces** screen to administer a node (C-LAN port or IP Interface port) as the default gateway.

The default node is a display-only entry on the **Node Names** screen with IP address 0.0.0.0. It acts as a variable that takes on unknown addresses as values. When the “default” IP route is set up, any address not known by the C-LAN is substituted for the default address in the default IP route, which uses the router as the default gateway.

Setting up Alternate Gatekeeper and C-LAN load balancing

Alternate Gatekeeper gives IP endpoints a list of available C-LAN circuit packs. Alternate Gatekeeper addresses and C-LAN load-balancing spread IP endpoint registration across more than one C-LAN circuit pack. The C-LAN load-balancing algorithm allocates endpoint registrations within a network region to the C-LAN with the least number of sockets in use. This increases system performance and reliability.

If registration with the original C-LAN circuit pack IP address is successful, the software sends back the IP addresses of all the C-LAN circuit packs in the same network region as the IP endpoint. If the network connection to one C-LAN circuit pack fails, the IP endpoint re-registers with a different C-LAN. If the system uses network regions based on IP address, the software also sends the IP addresses of C-LANs in interconnected regions. These alternate C-LAN addresses are also called *gatekeeper* addresses. These addresses can also be used if the data network carrying the call signaling from the original C-LAN circuit pack fails.

IP Telephones can be programmed to search for a gatekeeper independently of load-balancing. The IP Telephone accepts gatekeeper addresses in the message from the Dynamic Host Configuration Protocol (DHCP) server or in the script downloaded from the Trivial File Transfer Protocol (TFTP) server. If the phone cannot contact the first gatekeeper address, it uses an alternate address. If the extension and password is rejected by the first gatekeeper, the IP Telephone contacts the next gatekeeper. The number of gatekeeper addresses the phone accepts depends on the length of the addresses administered on the DHCP server.

Note:

A single Alternate Gatekeeper list is typically used in configurations with multiple media servers. In this case, the DHCP server sends the same Alternate Gatekeeper list to all IP endpoints, but a given IP endpoint may not be able to register with some of the gatekeepers in the list and a registration attempt to those gatekeepers will be rejected.

C-LAN load balancing and alternate gatekeeper addresses require IP stations that accept multiple IP addresses, such as:

- IP telephone
- IP Softphone
- Avaya IP Agent

Endpoint capabilities

Table 7: Endpoint capabilities

Endpoint	Number of Gatekeepers	How set
IP Telephone	1	Default - DNS name AvayaCallServer, or manually, one fixed IP address
	8	Through DHCP - DNS names or fixed IP addresses. DHCP limits all options to a total of 255 bytes.
	10	Through TFTP - DNS names or fixed IP addresses. TFTP overwrites any gatekeepers provided by DHCP
	30	Fixed IP addresses from Communication Manager. Communication Manager 2.0 and later supersedes any gatekeeper address provided previously.
IP Softphone R5	30	Manually through options or properties of the IP Softphone after it is installed.
IP Agent R3	30	Manually through options or properties of the IP agent after it is installed, or from Communication Manager.

Note:

DHCP servers send a list of alternate gatekeeper and C-LAN addresses to the IP Telephone endpoint. It is possible for a hacker to issue a false request and thereby obtain IP addresses from the DHCP server.However, the alternate gatekeeper IP addresses will only be sent to an endpoint that successfully registers.

TN802B MAPD (IP Interface Assembly)

The TN802 IP interface circuit pack supports voice calls and fax calls from the switch across a corporate intranet or the Internet. This circuit pack is still supported, but has been replaced with the TN2302AP IP Media Processor. The IP trunking software runs on an embedded PC that runs Windows NT. The TN802 circuit pack supports IP Solutions including IP trunking and MedPro (H.323) with IP Softphones.

The TN802 IP Interface operates in two modes:

- IP Trunk
- Media Processor (MedPro/H.323).

The TN802 defaults to IP Trunk mode. To use it in MedPro mode, you activate it by administration to use the H.323 trunking feature.

TN2302AP IP Media Processor

Use the TN2302AP IP Media Processor to transmit voice and fax data (non-DCS signaling) over IP connections, and for H.323 multimedia applications in H.323 V2 compliant endpoints.

The TN2302AP IP Media Processor provides port network connectivity on an IP-Connect configuration. The TN2302AP IP Media Processor includes a 10/100BaseT Ethernet interface to support H.323 endpoints for IP trunks and H.323 endpoints, and its design improves voice quality through its dynamic jitter buffers.

The TN2302AP IP Media Processor additionally performs the functions:

- Echo cancellation
- Silence suppression
- DTMF detection
- Conferencing

It supports the following codecs, fax detection for them, and conversion between them:

- G.711 (mu-law or a-law, 64Kbps)
- G.723.1 (6.3Kbps or 5.3Kbps audio)
- G.729 (8Kbps audio)

Improving theTN2302AP transmission interface

The TN2302AP IP Media Processor provides improved voice quality through its dynamic jitter buffers. The TN2302AP's digital signal processors (DSPs), by default, insert 5.0 dB of loss in the signal from the IP endpoints, and insert 5.0 dB of gain in the signal to the IP endpoints. System administrators can administer loss/gain, based on country code on the **terminal-parameters** screen.

Supporting TN2302AP hairpinning

The TN2302AP IP Media Processor supports 64 ports of shallow hairpin. IP packets that do not require speech codec transcoding can be looped back at the UDP/IP layers with a simple change of addressing. This reduces delay and leaves DSP resources available.

Testing TN2302AP ports

The TN2302AP IP Media Processor is a service circuit pack, not a trunk circuit pack. Therefore, an H.323 tie trunk cannot be used for facility test calls. Use the ping command to test the TN2302AP ports.

Enabling a survivable remote EPN

Any survivable remote EPN containing a C-LAN board and H.323 station sets should also contain a TN2302AP IP Media Processor.

TN2312BP IP Server Interface (IPSI)

In configurations with the S8700 Media Server controlling media gateways, the bearer paths and the control paths are separate. Control information for port networks (PNs) travels over a LAN through the Ethernet switch. The control information terminates on the S8700 Media Server at one end and on a TN2312BP IP Server Interface (IPSI) on the other end. Each IPSI may control up to five port networks by tunneling control messages over the Center-Stage or ATM network to PNs that do not have IPSIs.

Note:

IPSIs cannot be placed in a PN that has a Stratum-3 clock interface. Also, IPSIs cannot be placed in a remote PN that is using a DS1 converter.

In configurations that use a dedicated LAN for the control path, IPSI IP addresses are typically assigned automatically using DHCP service from the S8700. Also, a dedicated IPSI Ethernet connection to a laptop can be used to assign static IP addresses or for maintenance. In configurations using the customer’s LAN, only static addressing is supported.

Consult the *Avaya S8300, S8500, and S8700 Media Server Library* CD (555-233-825) for information on installing and upgrading S8700 and IPSI configurations.

MM760 VoIP Media Module

The Avaya MM760 Media Module is a clone of the motherboard VoIP engine.The MM760 provides the audio bearer channels for voice over IP calls, and is under control of the G700. Based on system administration of audio codecs, a MM760 can handle either 64 or 32 simultaneous channels of H.323 audio processing. If the IP Parameters form specifies only G.711 mu-law or G.711 a-law as the audio codecs, the MM760 can service 64 channels. If any other codec type (G.723-5.3K, G.723-6.3K, or G.729) is administered, the MM760 can only service 32 channels. These call types can be mixed on the same resource. In other words, the simultaneous call capacity of the resource is 64 G.711 Equivalent Calls.

Note:

The MM760 is not supported in the G350 Media Gateway.

Note:

Customers who want an essentially non-blocking system must add an additional MM760 Media Module, if they use more than two MM710 Media Modules in a single chassis. The additional MM760 provides an additional 64 channels.

What is the MM760 Ethernet interface

The MM760 must have its own Ethernet address. The MM760 requires a 10/100 Base T Ethernet interface to support H.323 endpoints for DEFINITY IP trunks and stations from another G700 Media Gateway.

Supporting voice compression on the MM760

The MM760 supports on-board resources for compression and decompression of voice for G.711 (A- and µ-law), G.729 and 729B, and G.723 (5.3K and 6.3K). The VoIP engine supports the following functionality:

- RTP and RTCP interfaces
- Dynamic jitter buffers
- DTMF detection
- Hybrid echo cancellation
- Silence suppression
- Comfort noise generation
- Packet loss concealment

The MM760 also supports transport of the following:

- Teletypewriter device (TTY) tone relay over the Internet
- Faxes over a corporate IP intranet

  Note:

  The path between endpoints for fax transmissions must use Avaya telecommunications and networking equipment.

  ⚠ **SECURITY ALERT:**

  Faxes sent to non-Avaya endpoints cannot be encrypted.

- Modem tones over a corporate IP intranet

  Note:

  The path between endpoints for modem tone transmissions must use Avaya telecommunications and networking equipment.

Administration of Avaya gateways

The following sections describe the administration of the Avaya gateways:

Adding a G600/G650 Media Gateway to an S8500 or S8700-series Media Server

Adding a G350, or G700 Media Gateway to an S8300 Media Server

Adding a G600/G650 Media Gateway to an S8500 or S8700-series Media Server

Note:

The procedure that follows is a synopsis of the essential network administration. Depending on the specific configuration, additional administration might be required. For more information, see the *Avaya S8300, S8500, and S8700 Media Server Library* CD (555-233-825).

What S8500, S8700-series attributes this procedure assumes

This procedure assumes that the Avaya Media Server:

- is installed and cabled
- is configured as a server
- is running Communication Manager
- has an administered IP Server Interface (IPSI) circuit pack with
  - an IP address
  - a gateway address
  - a physical network connection

Note:

In order to successfully complete this procedure, you must have the IP address of each IPSI circuit pack. Use the **`list ip-interface`** command at the media server SAT interface to look up this information.

To add a G600 or G650 Media Gateway to an S8500 or S8700-series Media Server

1. Open a SAT session and locate IPSI circuit pack(s) with the **`list configuration all`** command.

   Ensure that you understand which IPSI circuit pack is supporting the gateway connection.

2. Check the previous cabinet administration with the **`list cabinet`** command.

3. Type **`add cabinet`** ***`n`*** and press **Enter**

   where ***n*** is the port network number associated with this cabinet (1-64). This number should be one number higher than the number of cabinets already administered.

4. Type the appropriate information in the following fields:

   - **Room**
   - **Floor**
   - **Building**

5. For each carrier (A-D) in the **Carrier** column, type or select **rmc-port** in the **Carrier Type** column.

   You are selecting a rack-mounted carrier port network for the G600 or G650.

6. Press **Enter** to save the changes.

7. Check the administration with the **`list cabinet`** command.

8. Ensure that the administration in Steps 3-5 are reflected in the report.

9. Type **`add ipserver-interface`** ***`PNnumber`*** and press **Enter**

   where ***`PNnumber`*** is the number of the corresponding port network.

10. Administer the fields on the **IP Interfaces** screen (Table 8: IP Interfaces fields on page 57).

Table 8: IP Interfaces fields ***1 of 2***

Field	Value	Comments
Enable QoS	**y**	You are implementing a QoS strategy. See Chapter 3: Network quality administration for more information on administering the additional fields that appear when QoS is enabled.
	n	You are not implementing a QoS strategy.
Primary IPSI		
Location		Physical address of the primary IPSI circuit pack
Host		IP address of the primary IPSI circuit pack that is supporting this connection. **Note:** If you are using DHCP, the **Host** and **DHCP ID** fields are set by the DHCP server.
		1 of 2

Table 8: IP Interfaces fields ***2 of 2***

Field	Value	Comments
DHCP ID		Set by DHCP server, if applicable.
Secondary IPSI		
Location		The physical address of the secondary IPSI circuit pack.
Host		IP address of the secondary IPSI circuit pack that is supporting this connection. **Note:** If you are using DHCP, the **Host** and **DHCP ID** fields are set by the DHCP server.
DHCP ID		Set by DHCP server, if applicable.

2 of 2

11. Check the administration with
    - **`display ipserver-interface`** ***`PNnumber`*** command
    - **`list cabinet`** command.
12. Type **`save translations`** and press **Enter**.

Adding a G350, or G700 Media Gateway to an S8300 Media Server

Note:

The procedure that follows is a synopsis of the essential network administration. Depending on the specific configuration, additional administration might be required. For more information, see the *Avaya S8300, S8500, and S8700 Media Server Library* CD (555-233-825).

What the G350, or G700 configurations provide

The Avaya G350, and G700 Media Gateways provide a highly flexible, scalable communications solution that can be incorporated in customer networks in two general configurations:

- Paired with an Avaya S8300 Media Server primary controller, it can serve as a standalone gateway within a small or medium sized office.
- Using a remote S8300, S8500, or S8700/S8710 Media Server for its primary controller, it may serve as a gateway in a large branch office networked through a WAN to a central location.

In both configurations, a G350, or G700 may be equipped with an S8300 configured as an LSP (Local Survivable Processor), which takes over call control if the connection to the server is lost.

What S8300 attributes this procedure assumes

This procedure assumes that the Avaya Media Server S8300 has the following attributes:

- Installed and cabled
- Configured as a server
- Running Communication Manager

Administering the G350, or G700 on an S8300 Media Server

In adding a G350, or G700 Media Gateway to an S8300 Media Server, use the following procedures:

- To look up the media gateway serial number
- To administer the media gateway controller
- To ensure that Communication Manager is running
- To add the media gateway
- To administer the IP interface (S8300 Only)

To look up the media gateway serial number

1. Using HyperTerminal, connect to the **Media Gateway Processor** (MGP) command line interface.
2. Type **`show system`** and press **Enter**.
3. Record the number in the **Serial No** field.

   The media gateway serial number is required by the media server to establish a connection.

   Note:

   The serial number is case-sensitive; record the serial number carefully.

To administer the media gateway controller

1. Using HyperTerminal, connect to the MGP command line interface.
2. Type **`set mgc list`** ***`135.122.xxx.xxx`***, ***`135.122.xxx.xxx`*** (2nd IP address is for multiple gateway controllers).
3. Type **`show mgc`** and press **Enter**.
4. Check that the configuration information is correct.
5. Type **`reset mgp`** and press **Enter** to reset the media gateway processor.

   Wait for the media gateway to come back up.
6. Type **`show system`** and press **Enter.**

To ensure that Communication Manager is running

1. Log into the media server by launching the **Maintenance Web interface**.
2. In the left-hand pane under the **Server** category select **Process Status**.

   The **Process Status** screen asks you for additional parameters.
3. In the **content** section, enable the **Summary** button.
4. In the **frequency** section, enable the **Display Once** button.
5. Click on **View**.

   The **View Process Status Results** screen shows the status of the server’s key components.
6. Verify that the **CommunicaMgr** field reads **UP**.

To add the media gateway

1. Establish a SAT session at the S8300 Media Server through either:
   - The command line interface using the `dsat` or `sat` command
   - Avaya Site Administration
2. When asked whether to suppress alarm origination, press **Enter** for the default (**y**) or type **y** (yes).
3. When prompted, enter the appropriate terminal type:

For the operating system	Choose the terminal type
Windows NT 4.0 or Win95	**NTT**
Windows 2000	**W2KTT**
Sun Workstations	**SUNT**

4. At the SAT, type **`add media-gateway`** ***`gateway number`*** and press **Enter** to display the **Media Gateway** form.
5. In the **Name** field type the name for the media gateway.
6. In the **Serial No.**field type the media gateway's case-sensitive serial number (see To look up the media gateway serial number).

   Note:

   The IP Address and MAC Address fields automatically populate once the G350, or G700 media gateway registers with the media server.
7. Press **Enter** to save the administration.
8. Check the administration with the **`list media-gateway`** command.
9. Verify that information in the **Number**, **Name**, **Serial No.**, and **IP Address** fields is correct, and that the **Reg?** field is *y*.

To administer the IP interface (S8300 Only)

1. Type **`list ip-interface`** and press **Enter** to display the IP Interfaces form.
2. If the PROCR (processor) interface has already been enabled (**ON** field is **y**), then cancel the command.

   If the processor port has not been enabled

   a. Set the **ON** field to **y**.

   b. Set the **Net Rgn** (network region) to **1**.
3. Press **Enter** to save the changes.
4. Type **`save translation`** and press **Enter**.

Administration of IP trunks

The following sections describe the administration of IP trunks:

Administering SIP trunks

Administering H.323 trunks

Administering SIP trunks

SIP is the Session Initiated Protocol, an endpoint-oriented messaging standard defined by the Internet Engineering Task Force (IETF). As implemented by Avaya for release 2.0 of Communication Manager, SIP "trunking" functionality will be available on any of the Linux-based media servers (S8300, S8500 or S8700-series). These media servers will function as Plain Old Telephone Service (POTS) gateways, and they will also support name/number delivery between and among the various non-SIP endpoints supported by Communication Manager (analog, DCP or H.323 stations and analog, digital or IP trunks), and new SIP-enabled endpoints, such as the Avaya 4602SIP Telephone. In addition to its calling capabilities, IP Softphone R5 and later also will include instant-messaging client software, which is a SIP-enabled application, while continuing its full support of the existing H.323 standard for call control.

For more information on SIP administration and usage, see *SIP Support in Avaya Communication Manager 2.0*, 555-245-206, and *Converged Communications Server Installation and Administration*, 555-245-705.

Administering H.323 trunks

H.323 trunks use an ITU-T IP standard for LAN-based multimedia telephone systems. IP-connected trunks allow trunk groups to be defined as ISDN-PRI-equivalent tie lines between switches over an IP network. Trunks that use IP connectivity reduce costs and simplify management.

Benefits include:

- Reduction in long distance voice and fax expenses
- Facilitation of global communications
- Full-function networks with data and voice convergence
- Network optimization by using the existing network resources

The TN2302AP enables H.323 trunk service using IP connectivity between an Avaya IP solution and another H.323 v2-compliant endpoint.

H.323 trunk groups can be configured as:

- Tie trunks supporting ISDN trunk features such as DCS+ and QSIG
- Generic tie-trunks permitting interconnection with other vendors' H.323 v2-compliant switches
- Direct-inward-dial (DID) type public trunks, providing access to the switch for unregistered users

This section cover:

Setting up H.323 trunks for administration

Administering H.323 trunks

Setting up H.323 trunks for administration

This section describes the preliminary administration steps needed to set up H.323 trunks. Before you can administer an H.323 trunk, perform the following tasks:

- Verifying customer options for H.323 trunking
- Administering C-LAN and IP Media Processor circuit packs (S8500/S8700-series)

  Note:

  These circuit packs are not required if your system has built-in Ethernet capabilities (S8300).

- Administering QoS parameters
- Assigning IP node names and IP addresses
- Defining IP interfaces
- Assigning link through Ethernet data module (S8500/S8700)
- Implementing Best Service Routing (optional)

Verifying customer options for H.323 trunking

Verify that H.323 trunking is set up correctly on the **system-parameters customer-options** screen. If any changes need to be made to fields on this screen, call your Avaya representative for more information.

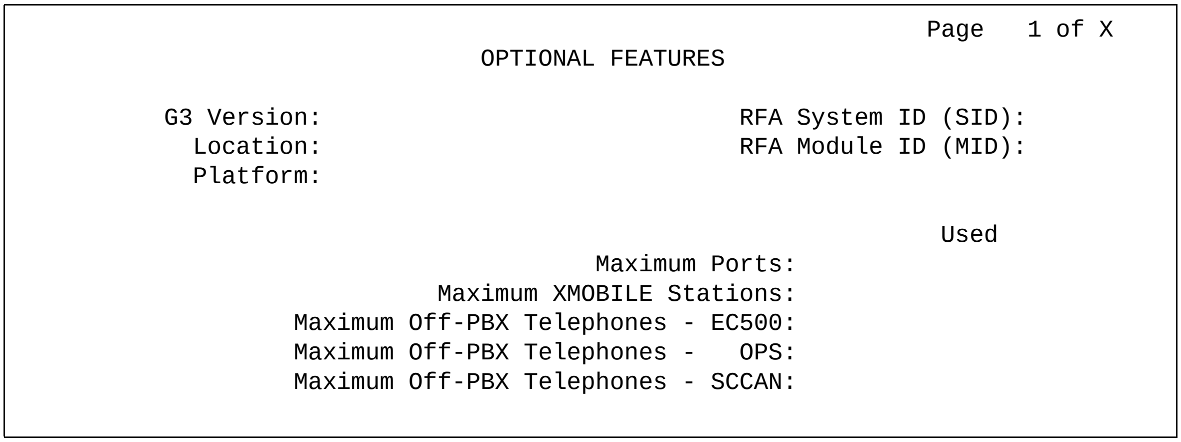
Note:

The **system-parameters customer-options** screen is display only. Use the **`display system-parameters customer-options`** command to review the screen. The License File controls the system software release, the Offer Category, features, and capacities. The *init* login does not have the ability to change the customer options, offer options, or special applications screens.

To verify customer options for H.323 trunking:

1. Type **`display system-parameters customer-options`**, and go to the **Optional Features** screen.

Optional Features screen



```
                                                               Page   1 of X
                                OPTIONAL FEATURES

          G3 Version:                            RFA System ID (SID):
            Location:                            RFA Module ID (MID):
            Platform:

                                                                Used
                                        Maximum Ports:
                             Maximum XMOBILE Stations:
                   Maximum Off-PBX Telephones - EC500:
                   Maximum Off-PBX Telephones -   OPS:
                   Maximum Off-PBX Telephones - SCCAN:
```

2. Verify that the following fields have been completed:

Field	Conditions/Comments
G3 Version	This value should reflect the current version of Communication Manager.
Maximum Administered H.323 Trunks	Number of trunks purchased. Value must be greater than 0.
Maximum Administered Remote Office Trunks	Number of remote office trunks purchased.

3. Go to the page that displays the **IP trunks** and **ISDN-PRI** fields.
4. Verify that **IP Trunks** and **ISDN-PRI** are enabled.

   If not, you need to obtain a new license file.

Administering C-LAN and IP Media Processor circuit packs (S8500/S8700-series)

To administer the C-LAN and IP Media Processor circuit packs:

1. Type **`change circuit-packs`** to open the **Circuit Packs** screen.

Circuit Packs screen

```
                            Page 2 of 5

                           Circuit Packs

Cabinet 1                                Carrier: B
                                    Carrier Type: port

Slot Code     SF Mode Name              Slot Code     SF Mode Name
00   TN799    C       C-LAN
01   TN2302           IP Media Processor
02
03
04
```

2. To administer a C-LAN circuit pack, complete the following fields:

Fields for CLAN	Conditions/Comments
Code	**TN799DP**
Name	**C-LAN (**displays automatically)

3. To administer an IP Media Processor, complete the following fields:

Fields for IP Media	Conditions/Comments
Code	**TN2302AP**
Name	**IP Media Processor** (displays automatically)

4. Submit the screen.

Administering QoS parameters

Four parameters on the **system-parameters maintenance** screen determine threshold Quality of Service (QoS) values for network performance. You can use the default values for these parameters, or you can change them to fit the needs of your network. (See Setting network performance thresholds).

Administer additional QoS parameters, including defining IP Network Regions and specifying the codec type to be used. See Chapter 3: Network quality administration.

Assigning IP node names and IP addresses

Communication Manager uses node names to reference IP addresses throughout the system. Use the **IP Node Names** screen to assign node names and IP addresses to each node in the network with which this switch communicates through IP connections. The **Node Names** screen must be administered on each node in an IP network.

A node can be:

- C-LAN Ethernet or PPP port
- Bridge or router
- CMS Ethernet port
- INTUITY AUDIX

Enter the AUDIX name and IP address on the **AUDIX Node Names** form. Enter data for all other node types on the **IP Node Names** screen.

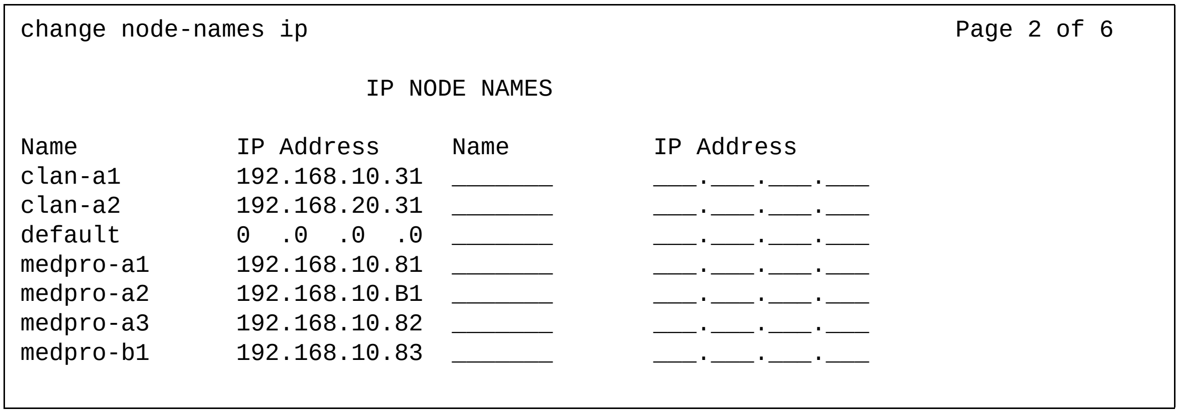
For H.323 connections, each MedPro Ethernet port (IP interface) on the local switch must also be assigned a node name and IP address on the **IP Node Names** screen.

Assign the node names and IP addresses in the network in a logical and consistent manner from the point of view of the whole network. Assign the names and addresses in the planning stages of the network and should be available from the customer system administrator or from an Avaya representative.

To assign IP Node Names:

1. Type **`change node-names ip`** to open the **IP Node Names** screen.

IP Node Names screen



```
change node-names ip                                             Page 2 of 6

                        IP NODE NAMES

Name           IP Address     Name          IP Address
clan-a1        192.168.10.31  _______       ___.___.___.___
clan-a2        192.168.20.31  _______       ___.___.___.___
default        0  .0  .0  .0  _______       ___.___.___.___
medpro-a1      192.168.10.81  _______       ___.___.___.___
medpro-a2      192.168.10.B1  _______       ___.___.___.___
medpro-a3      192.168.10.82  _______       ___.___.___.___
medpro-b1      192.168.10.83  _______       ___.___.___.___
```

2. Move to the fields below and complete them as follows:

Field	Conditions/Comments
Name	Enter unique node names for: ● Each C-LAN Ethernet port on the network ● Each IP Media Processor ● Each Remote Office ● Other IP gateways, hops, etc. The default node name and IP address is used to set up a default gateway, if desired. This entry is automatically present on the ***Node Names*** screen and cannot be removed. When the ***Node Names*** screen is saved, the system automatically alphabetizes the entries by node name.
IP Address	Enter a unique IP addresses for each node name.

3. Submit the screen.

Defining IP interfaces

The IP interface for each C-LAN and Media Processor circuit pack on the switch must be defined on the **IP Interfaces** screen. Each switch in an IP network has one **IP Interfaces** screen.

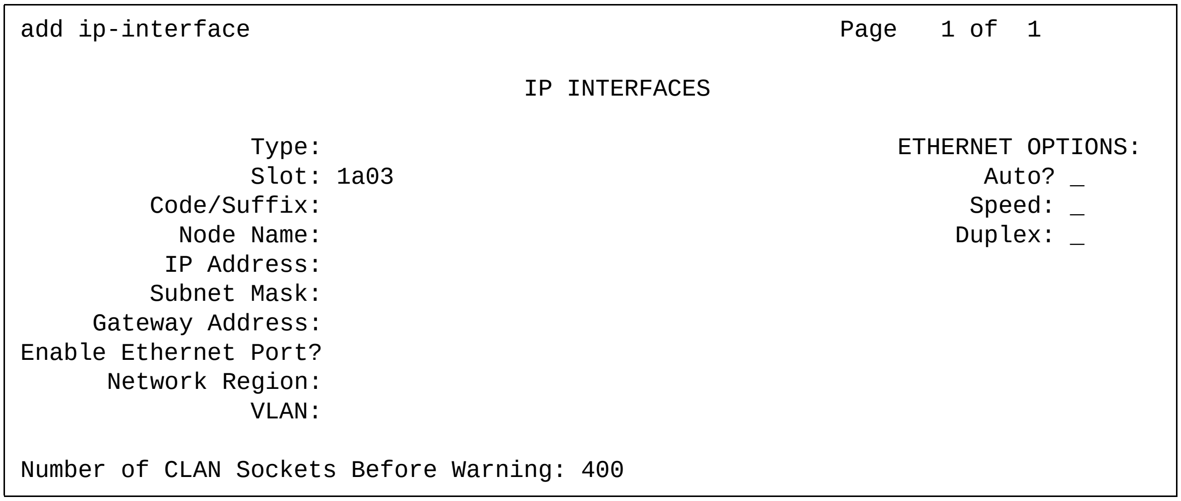
To define IP interfaces for each C-LAN and Media Processor circuit pack:

1. Type **`add ip-interface`** ***`CCccss`*** or ***`procr`*** to open the **IP Interfaces** screen.

Note:

This screen shows the display for the S8500/S8700 media servers.

IP Interfaces screen



```
add ip-interface                                         Page   1 of  1

                                   IP INTERFACES

                Type:                                        ETHERNET OPTIONS:
                Slot: 1a03                                         Auto? _
         Code/Suffix:                                             Speed: _
           Node Name:                                            Duplex: _
          IP Address:
         Subnet Mask:
     Gateway Address:
Enable Ethernet Port?
      Network Region:
                VLAN:

Number of CLAN Sockets Before Warning: 400
```

2. Complete the following fields as shown:

Field	Conditions/Comments
Type	Enter **C-LAN, MEDPRO,** or **PROCR**.
Slot	The slot location for the circuit pack.
Code/Suffix	Display only. This field is automatically populated with TN799DP for C-LAN or TN2302AP for IP Media Processor and the suffix letter(s).
Node name	The node name for the IP interface. This node name must already be administered on the IP Node Names screen.
IP Address	The IP address for this IP interface. The IP address is associated with the node name on the **Node Names** screen.
Subnet Mask	The subnet mask associated with the IP address for this IP interface.
Gateway Addr	The address of a network node that serves as the default gateway for the IP interface.
Enable Ethernet Port?	Enter **y**
Network Region	The region number for the IP interface. Enter a value between ● **1-80** (si only) ● **1-250** (S8300, S8500, or S8700)
VLAN	The 802.1Q virtual LAN value (**0 - 4094**) or **n** (no VLAN). This VLAN field interfaces with the TN799 (C-LAN) or TN802B Media Processor circuit packs; it does not send any instructions to IP endpoints.
Number of CLAN Sockets Before Warning	Always leave the default (**400**) unless instructed to enter a different value by Avaya Services.

3. Submit the screen.

Assigning link through Ethernet data module (S8500/S8700)

Note:

The S8300 does not support data modules.

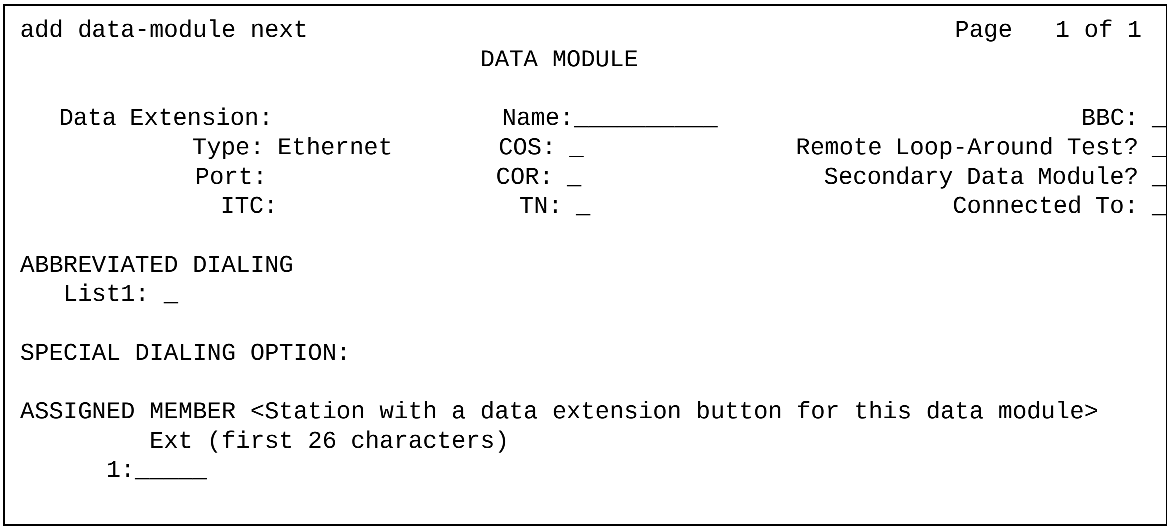
This section describes how to administer an Ethernet data module for the connection between the C-LAN circuit pack’s Ethernet port (port 17) and the LAN. The data module associates a link number and extension number with the C-LAN Ethernet port location. This association is used by the processor to set up and maintain signaling connections for multimedia call handling.

The C-LAN Ethernet port is indirectly associated with the C-LAN IP address through the slot location (which is part of the port location) on the **IP Interfaces** screen and the node name, which is on both the **IP Interfaces** and **Node Names** screens.

To assign a link through an Ethernet data module:

1. Type **`add data-module`** ***`next`*** to open the **Data Module** screen.

Data Module screen



```
add data-module next                                           Page   1 of 1
                                DATA MODULE

  Data Extension:               Name:__________                        BBC: _
           Type: Ethernet       COS: _             Remote Loop-Around Test? _
           Port:               COR: _               Secondary Data Module? _
             ITC:                TN: _                        Connected To: _

ABBREVIATED DIALING
   List1: _

SPECIAL DIALING OPTION:

ASSIGNED MEMBER <Station with a data extension button for this data module>
         Ext (first 26 characters)
      1:_____
```

2. Complete the following fields as shown:

Field	Conditions/Comments
Data Extension	Populated automatically with the **next** qualifier or type the extension number.
Type	Enter **Ethernet**. This indicates the data-module type for this link.
Port	Ethernet connections must be assigned to port **17** on the C-LAN circuit pack.
Link	Enter the link number, a link not previously assigned on this switch.
Name	Display only. The name appears in lists generated by the **`list data module`** command.
Network uses 1's for broadcast addresses	Enter **y** if the private network contains only Avaya switches and adjuncts. Enter **n** if the network includes non-Avaya switches that use the 0's method of forming broadcast addresses.

3. Submit the screen.

Implementing Best Service Routing (optional)

Use H.323 trunks to implement Best Service Routing (BSR). You can use H.323 trunks for polling, or for both polling and interflow. Because polling requires only a small amount of data exchange, the additional network traffic is insignificant. However, interflow requires a significant amount of bandwidth to carry the voice data. Depending on the other uses of the LAN/WAN and its overall utilization rate, voice quality could be degraded to unacceptable levels.

Avaya recommends that if H.323 trunks are used for BSR interflow, the traffic should be routed to a low-occupancy or unshared LAN/WAN segment. Alternatively, you might want to route internal interflow traffic, which may have lower quality-of-service requirements, over H.323 trunks, and route customer interflow traffic over circuit-switched tie trunks.

Administering H.323 trunks

You have completed the pre-administration tasks to set up H.323 trunks (see Setting up H.323 trunks for administration). This section describes the tasks that you need to complete to administer an H.323 trunk. Sample values are used to populate the fields to show the relationships between the screens and fields. Perform the following tasks:

- Creating an H323 trunk signaling group

  Create a signaling group for the H.323 trunks that connect this switch to a far-end switch.
- Creating a trunk group for H.323 trunks
- Modifying the H.323 trunk signaling group

  Modify the signaling group by entering the H.323 trunk group number in the **Trunk Group for the Channel Selection** field of the **Signaling Group** screen.

Creating an H323 trunk signaling group

Create a signaling group that is associated with H.323 trunks that connect this switch to a far-end switch. One or more unique signaling groups must be established for each far-end node to which this switch is connected through H.323 trunks.

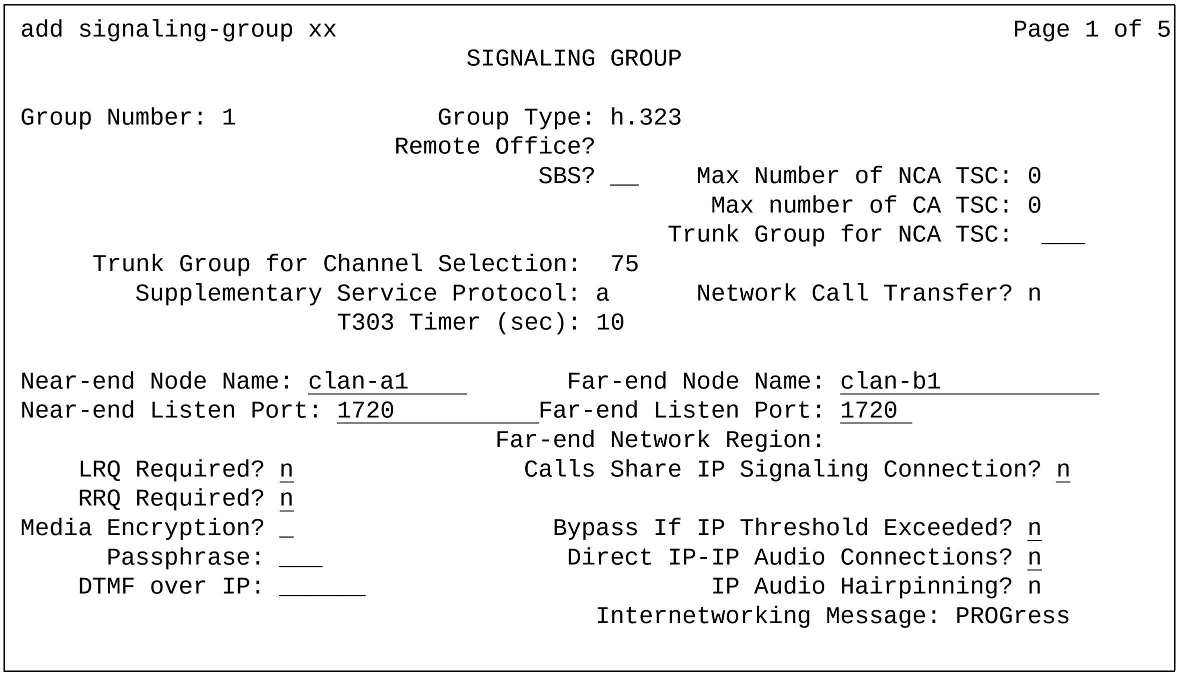
Note:

The following steps address only those fields that are specifically related to H.323 trunks. The other fields are described in the *Administrator's Guide for Avaya Communication Manager, 555-233-506*.

To create an H.323 trunk signaling group, do the following:

1. Type **`add signaling-group`** ***`number`*** to open the **Signaling Group** screen.

Signaling Group screen



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

Group Number: 1             Group Type: h.323
                         Remote Office?
                                   SBS? __    Max Number of NCA TSC: 0
                                               Max number of CA TSC: 0
                                            Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  75
        Supplementary Service Protocol: a      Network Call Transfer? n
                      T303 Timer (sec): 10

Near-end Node Name: clan-a1            Far-end Node Name: clan-b1
Near-end Listen Port: 1720            Far-end Listen Port: 1720
                                 Far-end Network Region:
    LRQ Required? n               Calls Share IP Signaling Connection? n
    RRQ Required? n
Media Encryption? _                 Bypass If IP Threshold Exceeded? n
      Passphrase: ___                Direct IP-IP Audio Connections? n
    DTMF over IP: ______                       IP Audio Hairpinning? n
                                        Internetworking Message: PROGress
```

2. Complete the following fields as shown:

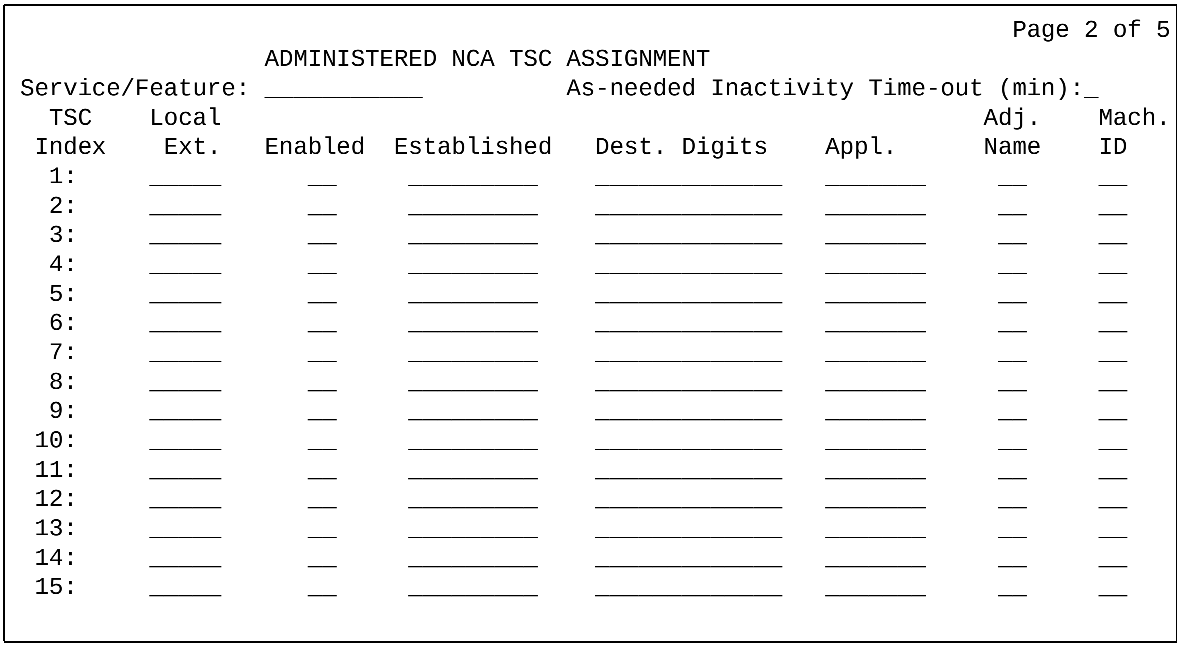
Table 9: Signaling Group screen options ***1 of 2***

Field	Conditions/Comments
Group Type	Enter **h.323**
Trunk Group for Channel Selection	Leave blank until you create a trunk group in the following task, then use the change command and enter the trunk group number in this field.
T303 Timer	Use this field to enter the number of seconds the system waits for a response from the far end before invoking Look Ahead Routing. Appears when the Group Type field is isdn-pri (DS1 Circuit Pack screen) or h.323 (Signaling Group screen).
Near-end Node Name	Enter the node name for the C-LAN IP interface on this switch. The node name must be administered on the **Node Names** screen and the **IP Interfaces** screen.
	1 of 2

Table 9: Signaling Group screen options ***2 of 2***

Field	Conditions/Comments
Far-end Node Name	This is the node name for the far-end C-LAN IP Interface used for trunks assigned to this signaling group. The node name must be administered on the **Node Names** screen on this switch. Leave blank when the signaling group is associated with an unspecified destination.
Near-end Listen Port	Enter an unused port number from the range **1719**, **1720** or **5000–9999**. Avaya recommends **1720**. If the **LRQ** field is **y**, enter **1719**.
Far-end Listen Port	Enter the same number as the one in the **Near-end Listen Port** field. This number must match the number entered in the **Near-end Listen Port** field on the signaling group screen for the far-end switch. Leave blank when the signaling group is associated with an unspecified destination.
Far-end Network Region	Identify network assigned to the far end of the trunk group. The region is used to obtain the codec set used for negotiation of trunk bearer capability. If specified, this region is used instead of the default region (obtained from the C-LAN used by the signaling group) for selection of a codec. Enter a value between **1-44**. Leave blank to select the region of the near-end node (C-LAN).
LRQ Required	Enter **n** when the far-end switch is an Avaya product. Enter **y** when the far-end switch requires a location request to obtain a signaling address in its signaling protocol.
Calls Share IP Signaling Connection	Enter **y** for connections between Avaya equipment. Enter **n** when the local and/or remote switch is not Avaya's.
RRQ Required	Enter **y** when a vendor registration request is required.
Bypass if IP Threshold Exceeded	Enter **y** to automatically remove from service trunks assigned to this signaling group when IP transport performance falls below limits administered on the **Maintenance-Related System Parameters** screen.
	2 of 2

3. If using DCS, go to the **Administered NCA TSC Assignment** page of this screen.



```
                                                                     Page 2 of 5
                 ADMINISTERED NCA TSC ASSIGNMENT
Service/Feature: ___________          As-needed Inactivity Time-out (min):_
  TSC    Local                                                     Adj.    Mach.
 Index    Ext.   Enabled  Established   Dest. Digits    Appl.      Name    ID
  1:     _____      __     _________    _____________   _______     __     __
  2:     _____      __     _________    _____________   _______     __     __
  3:     _____      __     _________    _____________   _______     __     __
  4:     _____      __     _________    _____________   _______     __     __
  5:     _____      __     _________    _____________   _______     __     __
  6:     _____      __     _________    _____________   _______     __     __
  7:     _____      __     _________    _____________   _______     __     __
  8:     _____      __     _________    _____________   _______     __     __
  9:     _____      __     _________    _____________   _______     __     __
 10:     _____      __     _________    _____________   _______     __     __
 11:     _____      __     _________    _____________   _______     __     __
 12:     _____      __     _________    _____________   _______     __     __
 13:     _____      __     _________    _____________   _______     __     __
 14:     _____      __     _________    _____________   _______     __     __
 15:     _____      __     _________    _____________   _______     __     __
```

4. Enter NCA TSC information on this screen.
5. Submit the screen.

Creating a trunk group for H.323 trunks

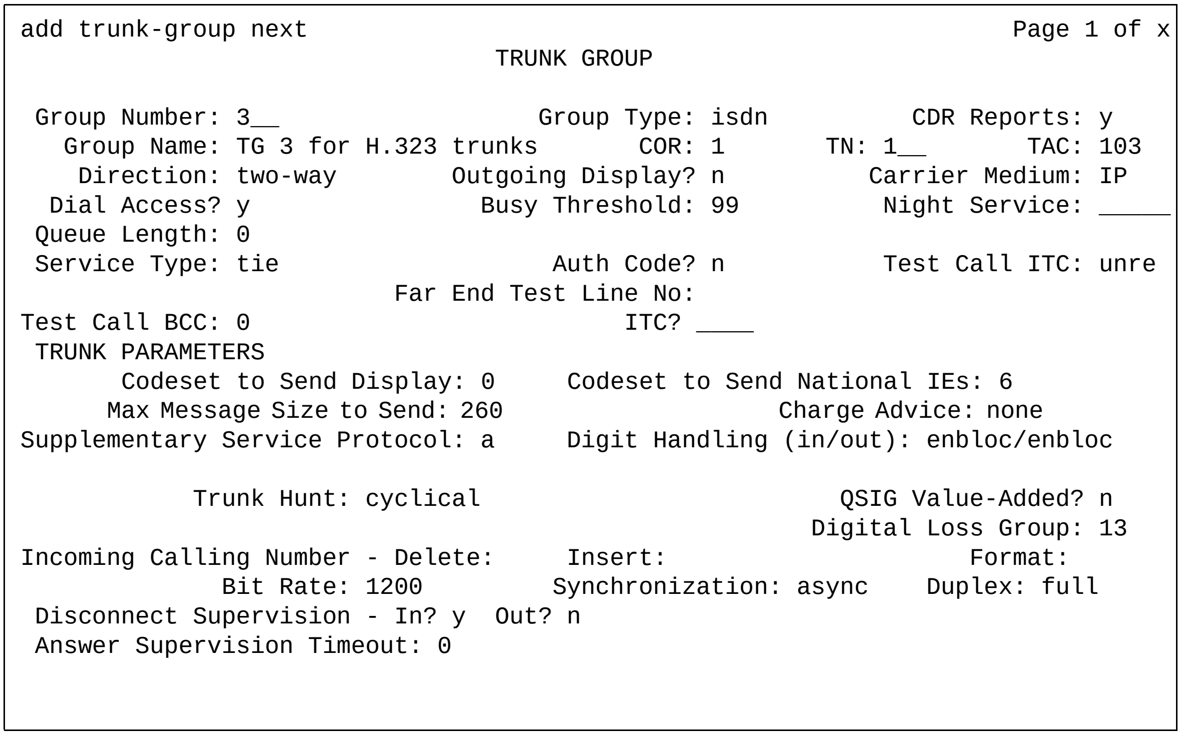
This task creates a new trunk group for H.323 trunks. Each H.323 trunk must be a member of an ISDN trunk group and must be associated with an H.323 signaling group.

> **Note:**
>
> The following steps address only those fields that are specifically related to H.323 trunks. The other fields are described in the *Administrator's Guide for Avaya Communication Manager, 555-233-506*.

To create an H.323 trunk group, do the following:

1. Type **`add trunk-group`** ***`next`*** to open the **Trunk Group** screen.

Trunk Group screen



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

 Group Number: 3__                 Group Type: isdn          CDR Reports: y
   Group Name: TG 3 for H.323 trunks       COR: 1       TN: 1__       TAC: 103
    Direction: two-way        Outgoing Display? n         Carrier Medium: IP
  Dial Access? y                Busy Threshold: 99         Night Service: _____
 Queue Length: 0
 Service Type: tie                  Auth Code? n           Test Call ITC: unre
                          Far End Test Line No:
Test Call BCC: 0                         ITC? ____
 TRUNK PARAMETERS
       Codeset to Send Display: 0     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. Complete the following fields as shown:

Field	Conditions/Comments
Group Type	Enter **isdn**
Carrier Medium	Enter **ip**
Service Type	Enter **tie**
TestCall ITC	Enter **unre** (unrestricted).
TestCall BCC	Enter **0**
Codeset to Send Display	Enter **0**
Outgoing Display	This field may need to be changed if the far-end is not Avaya's.

3. If using DCS, go to the **Trunk Features** page of this screen.

Trunk Features screen

```
add trunk-group next                                          Page   2 of  x
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member:
                                      Send Name: n      Send Calling Number: n
            Used for DCS? y
   Suppress # Outpulsing? n    Format: public
 Outgoing Channel ID Encoding: exclusive     UUI IE Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n
Network Call Redirection: none                Modify Tandem Calling Number? n
             Send UUI IE? y
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                          DS1 Echo Cancellation? n

                                          US NI Delayed Calling Name Update? n

                   SBS? n   Network (Japan) Needs Connect Before Disconnect? n
DSN Term? n
```

4. Complete the following fields as shown:

Field	Conditions/Comments
Used for DCS	Enter **y**.
Send Name Send Calling Number	These fields may need to be changed if the far-end is not Avaya’s.

5. To add a second signaling group, go to the **Group Member Assignments** page of this screen.

```
add trunk-group next                                       Page 6 of  x
                                 TRUNK GROUP
                                           Administered Members (min/max):   0/0
GROUP MEMBER ASSIGNMENTS                       Total Administered Members:   0
                                                                          Ans.
       Port    Code Sfx Name        Night           Mode      Type      Delay
  1: ip        H.323 Tr 1                            __       ___        ___
  2: ip        H.323 Tr 2                            __       ___        ___
  3: ip        H.323 Tr 3                            __       ___        ___
  4:
  5:
```

Note:

Each signaling group can support up to 31 trunks. If you need more than 31 trunks between the same two switches, add a second signaling group with different listen ports and add a second trunk group.

6. Enter group numbers using the following fields:

Field	Conditions/Comments
Port	Enter **ip**. When the screen is submitted, this value is automatically changed to a **T** number (**Txxxxx**).
Name	Enter a 10-character name to identify the trunk.
Mode	This field specifies the signaling mode used on tie trunks with TN722A or later, TN760B or later, TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs. This entry must correspond to associated dip-switch settings on the circuit pack.
Type	The **Type** column appears when the **Trunk Type** field is blank or **cont**. The **Type** column does not display if the **Trunk Type** field is **dis**.
Ans. Delay	**20** to **5100** in increments of 20: Specifies the length of time (in ms) your server running Communication Manager will wait before it sends answer supervision for incoming calls on tie trunks using the TN722A or later, TN760 (B, C, or D), TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs. Blank: Same as setting the field to **0**.

⚠ **CAUTION:**

Customers should not attempt to administer the last three fields. Please contact your Avaya representative for assistance.

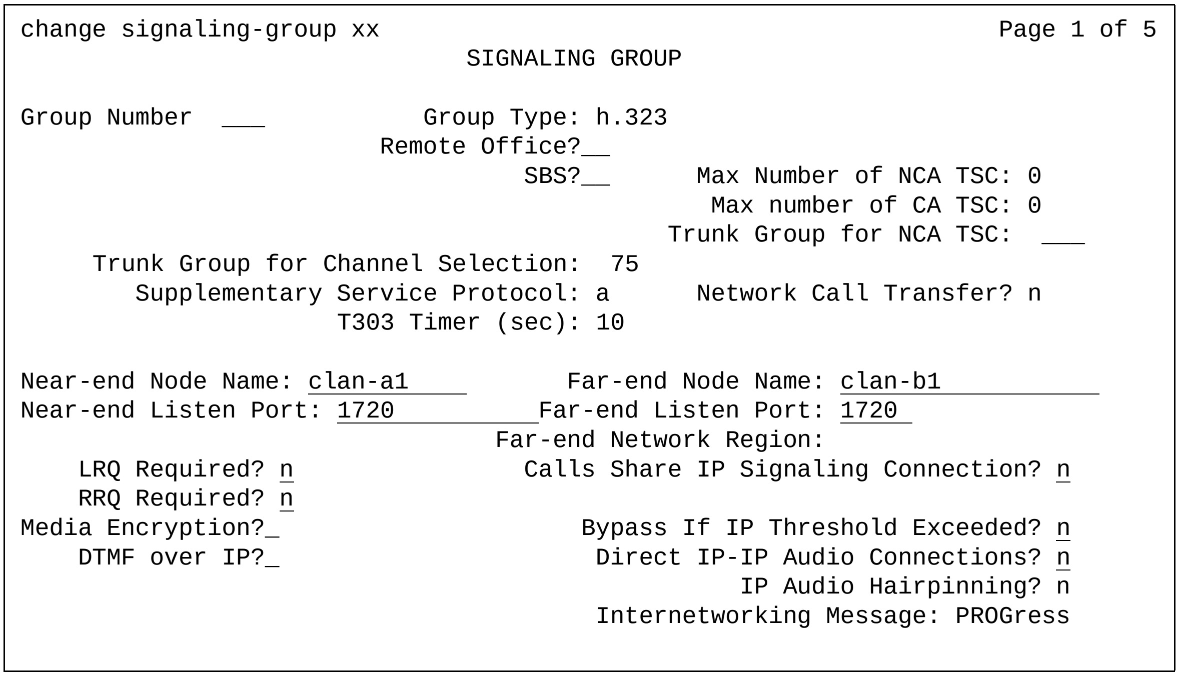
Modifying the H.323 trunk signaling group

Modify the **Signaling Group** screen to add a trunk group number to the **Trunk Group for Channel Selection** field.

To modify an H.323 trunk signaling group:

1. Type **`busy signaling-group`** ***`number`*** to busy-out the signaling group.
2. Type **`change signaling-group`** ***`number`*** to open the **Signaling Group** screen.

Signaling Group screen



```
change signaling-group xx                                           Page 1 of 5
                               SIGNALING GROUP

Group Number  ___           Group Type: h.323
                         Remote Office?__
                                   SBS?__      Max Number of NCA TSC: 0
                                                Max number of CA TSC: 0
                                             Trunk Group for NCA TSC:  ___
     Trunk Group for Channel Selection:  75
        Supplementary Service Protocol: a      Network Call Transfer? n
                      T303 Timer (sec): 10

Near-end Node Name: clan-a1           Far-end Node Name: clan-b1
Near-end Listen Port: 1720          Far-end Listen Port: 1720
                                 Far-end Network Region:
    LRQ Required? n                Calls Share IP Signaling Connection? n
    RRQ Required? n
Media Encryption?_                     Bypass If IP Threshold Exceeded? n
    DTMF over IP?_                      Direct IP-IP Audio Connections? n
                                                  IP Audio Hairpinning? n
                                        Internetworking Message: PROGress
```

3. Complete the following field:

Field	Conditions/Comments
Trunk Group for Channel Selection	Enter the trunk group number. If there is more than one trunk group assigned to this signaling group, the group entered in this field is the group that accepts incoming calls.

4. Submit the screen.

5. Type **`release signaling-group`** ***`number`*** to release the signaling group.

Administration of Avaya phones

The following sections describe the installation and administration of Avaya IP telephones:

- Administering IP Softphones
- Installing and administering Avaya IP telephones

Administering IP Softphones

IP Softphones operate on a PC equipped with Microsoft Windows and with TCP/IP connectivity through Communication Manager. Avaya offers three different Softphone applications:

- IP Softphone for any phone user
- IP Agent for call center agents
- Softconsole for attendants

IP Softphones can be configured to operate in any of the following modes:

- **Road-warrior** mode consists of a PC running the Avaya IP Softphone application and Avaya iClarity IP Audio, with a single IP connection to an Avaya server or gateway.
- **Telecommuter** mode consists of a PC running the Avaya IP Softphone application with an IP connection to the server, and a standard telephone with a separate PSTN connection to the server.
- **Shared Control** mode provides a registration endpoint configuration that will allow an IP Softphone and a non-Softphone telephone to be in service on the same extension at the same time. In this new configuration, the call control is provided by both the Softphone and the telephone endpoint. The audio is provided by the telephone endpoint.

Documentation on how to set up and use the IP Softphones is included on the CD-ROM containing the IP Softphone software. Procedures for administering Communication Manager to support IP Softphones are given in *Administrator's Guide for Avaya Communication Manager, 555-233-506.*

Administering the IP Softphone

This section focuses on administration for the trunk side of the Avaya IP Solutions offer, plus a brief checklist of IP Softphone administration. Comprehensive information on the administration of IP Softphones is given in *Administrator's Guide for Avaya Communication Manager, 555-233-506*.

There are two main types of IP Softphone configurations:

- Administering a Telecommuter phone
- Administering a Road-warrior phone

Communication Manager can distinguish between various IP stations at RAS using the product ID and release number sent during registration. An IP phone with an Avaya manufacturer ID can register if the number of stations with the same product ID and the same or lower release number *is less than* the administered system capacity limits. System limits are based on the number of simultaneous registrations.

Administering a Telecommuter phone

The Telecommuter uses two connections: one to the PC over the IP network and another connection to the telephone over the PSTN. The user places and receives calls with the IP Softphone interface running on a PC and uses the telephone handset to speak and listen.

Note:

> The **System Parameters Customer Options** screen is display only. Use the **`display system-parameters customer-options`** command to review the screen. The License File controls the system software release, the Offer Category, features, and capacities. The *init* login does not have the ability to change the customer options, offer options, or special applications screens.

To administer a Telecommuter phone:

1. Type **`display system-parameters customer-options`** and press **Enter** to open the **System Parameters Customer Options** screen.

   Verify that IP Softphone is enabled. Review the following fields on the screen:

Field	**Value**
Maximum Concurrently Registered IP Stations	Identifies the maximum number of IP stations that are simultaneously registered, not the maximum number that are simultaneously administered. This value must be greater than **0**, and must be less than or equal to the value for Maximum Ports.
Maximum Concurrently Registered Remote Office Stations	Specifies the maximum number of remote office stations that are simultaneously registered, not the maximum number that are simultaneously administered. This value must be greater than **0**, and must be less than or equal to the value for Maximum Ports.
IP Stations	This value should be **y.**
Product ID	This is a 10-character field that allows any character string. For new installations, IP Soft, IP Phone, IP Agent and IP ROMax, the product IDs automatically appear
Rel. (Release)	Identifies the release number.
Limit	This field defaults to the maximum allowed value, based on the **Concurrently Registered Remote Office Stations** field on page 1 of the ***System Parameters Customer Options*** screen.

2. Type **`add station`** ***`next`*** and press **Enter** to open the **Station** screen and complete the fields listed in the table below to add a DCP station (or change an existing DCP station):

Field	**Value**
Type	Enter the phone model, such as **6408D**.
Port	Enter **x** if virtual, or the port number of an existing phone.
Security Code	Enter the user's password.
IP Softphone	Enter **y**.

3. Go to page 2; verify whether the field **Service Link Mode:** *`as-needed`* is set as shown.

4. Install the IP Softphone software on the user's PC.

Administering a Road-warrior phone

The road-warrior uses two separate software applications running on a PC that is connected over an IP network. The single network connection carries two channels: one for call control signaling and one for voice. IP Softphone software handles the call signaling. With IP Softphone R5 or greater, iClarity is automatically installed to handle voice communications.

Note:

> The **System Parameters Customer Options** screen is display only. Use the **`display system-parameters customer-options`** command to review the screen. The License File controls the system software release, the Offer Category, features, and capacities. The *init* login does not have the ability to change the customer options, offer options, or special applications screens.

To administer a Road-warrior phone:

1. Type **`display system-parameters customer-options`**.

   Verify that IP Softphone is enabled. Go to the appropriate pages on the **System Parameters Customer Options** screen to review the following fields:

Field	**Value**
Maximum Concurrently Registered IP Stations	Specifies the maximum number of IP stations that are simultaneously registered, not the maximum number that are simultaneously administered. This value must be greater than **0**.
IP Stations	Must be **y**.

Field	Value
Product ID	This is a 10-character field that allows any character string. For new installations, IP Soft, IP Phone, IP Agent and IP ROMax product IDs automatically display.
Rel. (Release)	Identifies the release number
Limit	Defaults to **1**

2. If this is for a dual-connect IP Softphone (R2 or earlier), go to the **Station** screen and complete the fields listed in the table below to add an H.323 station:

Field	Value
Type	Enter **H.323**.
Port	Enter **x**.

3. Type **`add station`** ***`next`*** and press **Enter** to open the **Station** screen and complete the fields listed in the table below to add a DCP station (or change an existing DCP station):

Field	Value
Type	Enter the phone model you wish to use, such as **6408D**.
Port	Enter **x** if virtual, or the port number of an existing phone. If only an IP Softphone, enter **IP**.
Security Code	Enter the user’s password.
IP Softphone	Enter **y**.

4. Go to page 2; **Service Link Mode:** *`as-needed`*.

   Install the IP Softphone software on the user’s PC (iClarity automatically installed with the IP Softphone R2 or greater).

5. For pre-R2 IP Softphones, an H.323 V2-compliant audio application (such as Microsoft NetMeeting) must be installed.

Installing and administering Avaya IP telephones

The Avaya line of digital business phones uses Internet Protocol (IP) technology with Ethernet line interfaces and has downloadable firmware.

IP Telephones provide support for dynamic host configuration protocol (DHCP) and trivial file transfer protocol (TFTP) over IPv4/UDP, which enhance the administration and servicing of the phones.

For more information on installing and administering Avaya IP telephones, see *4600 Series IP Telephone R2.1 LAN Administrator's Guide*, 555-233-507.

About the 46xx IP telephone series

The 46xx IP Telephone product line possesses a number of shared model features and capabilities. All models also feature

- Downloadable firmware
- Automatic IP address resolution through DHCP
- Manual IP address programming.

The 46xx IP Telephone product line includes the following telephones:

- Avaya 4601 IP telephone
- Avaya 4602 IP telephone
- 4602SIP IP telephone
- Avaya 4602SW IP telephone
- Avaya 4606 IP telephone
- Avaya 4610SW IP telephone
- Avaya 4620 IP telephone
- Avaya 4620SW IP telephone
- Avaya 4624 IP telephone
- Avaya 4630SW IP Screenphone
- Avaya 4690 IP conference telephone

For information on the feature functionality of the IP telephones, see the *Hardware Guide for Avaya Communication Manager (555-245-207)*, the *4600 Series IP Telephone Installation Guide (555-233-128)*, or the appropriate 4600 series IP Telephone user's guide.

About IP telephone hardware/software requirements

Note:

Communication Manager requires that IP telephones still running R1.1 or earlier software be upgraded to R1.51 or newer software. The earlier software used a dual connection architecture that is no longer supported.

46xx IP Telephones are shipped from the factory with operational firmware installed. Some system-specific software applications are downloaded from a TFTP server through automatic power-up or reset. The 46xx IP Telephones search and download new firmware from the TFTP server before attempting to register with the Communication Manager.

The software treats the 46xx IP Telephones as any new station type, including the capability to **`list/display/change/duplicate/alias/remove station`**.

Note:

Audio capability for the IP Telephones requires the presence of the TN2302AP IP Media Processor circuit pack, which provides hairpinning and IP-IP direct connections. This conserves TDM bus and timeslot resources and improves voice quality.

The 46xx IP Telephone also requires a TN799DP Control-LAN (C-LAN) circuit pack for the signaling capability on the DEFINITY Server csi and si platforms. You do not need a C-LAN circuit pack to connect an IP Telephone if your system has built-in Ethernet capability (for example Avaya S8300 Media Server or Avaya S8700 Media Server).

To install required TN2302AP and TN799DP circuit Packs, if necessary

1. Determine the carrier/slot assignments of the circuit packs to be added.
2. Insert the circuit pack into the slot specified in step 1.

Note:

You do not have to power down the cabinet to install the circuit packs.

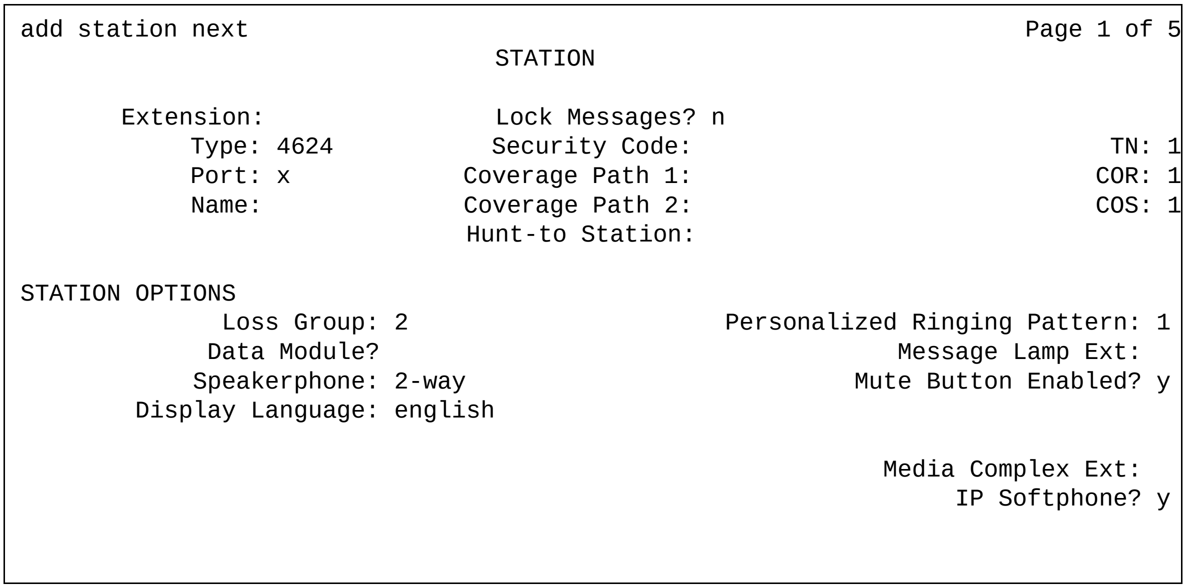
Administering Avaya IP telephones

IP Telephones R1.5 or greater use a single connection, and you only need to administer the station type.

To add an IP telephone

1. Type **`add station`** ***`next`*** to go to the **Station** screen.

Station screen



```
add station next                                                      Page 1 of 5
                                 STATION

       Extension:                Lock Messages? n
            Type: 4624           Security Code:                             TN: 1
            Port: x            Coverage Path 1:                            COR: 1
            Name:              Coverage Path 2:                            COS: 1
                               Hunt-to Station:

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

                                                            Media Complex Ext:
                                                                 IP Softphone? y
```

2. Complete the fields as shown in the following table:

Field	Value
Extension Type	Enter the IP Telephone 46 model number such as **4624**. The following phones are administered with an alias: ● 4601 (administer as a 4602) ● 4602SW (administer as a 4602) ● 4690 (administer as a 4620)
Port	Enter **x**, or **IP**.

Note:

A 46xx IP Telephone is always administered as an X port, and then once it is successfully registered by the system, a virtual port number is assigned.

3. For dual-connection architecture IP Telephones (R2 or earlier), complete the fields as shown in the following table:

Field	Value
Media Complex Ext	Enter the H.323 administered extension.
Port	Enter **x**.

4. Submit the screen.

About hairpinning and shuffling

Avaya Communication Manager can shuffle or hairpin call path connections between two IP endpoints by rerouting the voice channel away from the usual TDM bus connection and creating a direct IP-to-IP connection. Shuffling and hairpinning are similar because they preserve connection and conversion resources that might not be needed, depending on the compatibility of the endpoints that are attempting to interconnect.

Shuffling and hairpinning techniques differ in the way that they bypass the unnecessary call-path resources (compare either Figure 4: Shuffled audio connection between IP endpoints in the same network region on page 87 or Figure 5: Shuffled audio connection between IP endpoints in different network regions on page 88 with Figure 6: Hairpinned audio connection between 2 IP endpoints in the same network region on page 91).

Shuffled or hairpinned connections:

- Conserve channels on the TN2302AP IP Media Processor.
- Bypass the TDM bus, conserving timeslots.
- Improve voice quality by bypassing the codec on the TN2302AP IP Media Processor circuit pack.

Because shuffling frees up more resources on the TN2302AP IP Media Processor circuit pack than hairpinning does, Communication Manager first checks both endpoints to determine whether the Determining if shuffling is possible on page 86 are met. If the shuffling criteria are not met, Communication Manager routes the call according to the What are the criteria for hairpinning on page 90, if hairpinning is enabled. If hairpinning is not enabled, Communication Manager routes the call to the TDM bus. Both endpoints must connect through the same TN2302AP IP Media Processor for Communication Manager to shuffle or hairpin the audio connection.

For information on interdependencies that enable hairpinning and shuffling audio connections, see Hairpinning and shuffling administration interdependencies on page 92. For a discussion of Network Address Translation (NAT), see About Network Address Translation (NAT) on page 93.

Note:

See Chapter 5: Feature interactions and considerations for feature interaction information and other considerations for using shuffling and hairpinning.

What hardware and endpoints are required

The TN2302AP IP Media Processor circuit pack is required for shuffling or hairpinning audio connections.

The specific endpoint types that you can administer for hairpinning or shuffling are:

- All Avaya IP stations
- Other vendors' H.323-compatible stations

What are shuffled audio connections

Shuffling an audio connection between two IP endpoints means rerouting voice channel away from the usual TDM bus connection and creating a direct IP-to-IP connection. Shuffling saves such resources as TN2302AP channels and TDM bus time slots and improves voice quality because the shuffled connection bypasses the TN2302AP's codec. Both endpoints must be capable of shuffling (support H.245 protocol) before Communication Manager can shuffle a call.

Determining if shuffling is possible

Communication Manager uses the following criteria to determine whether a shuffled audio connection is possible:

- A point-to-point voice connection exists between two endpoints.
- No other active call (in-use or held) that requires TDM connectivity (for example, applying tones, announcement, conferencing, and others) exists on either endpoint.
- The endpoints are in the same network region or in different, interconnected regions.
- Both endpoints or connection segments are administered for shuffling by setting the **Direct IP-IP Audio Connections** field on the Station screen on page 104 or the Signaling group screen on page 103) to **y**.
- If the **Direct IP-IP Audio Connections** field is **y** (yes), but during registration the endpoint indicates that it does not support audio shuffling, then a call cannot be shuffled.

  If the **Direct IP-IP Audio Connections** field is **n** (no), but during registration the endpoint indicates that it can support audio shuffling, then calls to that endpoint cannot be shuffled, giving precedence to the endpoint administration.
- The rules for Inter-network region connection management on page 99 are met.
- There is at least one common codec between the endpoints involved and the Inter-network region Connection Management codec list.
- The endpoints have at least one codec in common as shown in their current codec negotiations between the endpoint and the switch.
- Both endpoints can connect through the same TN2302AP IP Media Processor circuit pack.

What are shuffling examples

Shuffling within the same network region

Figure 4: Shuffled audio connection between IP endpoints in the same network region on page 87 and Figure 5: Shuffled audio connection between IP endpoints in different network regions on page 88 provide examples of shuffled audio connections.

Figure 4: Shuffled audio connection between IP endpoints in the same network region

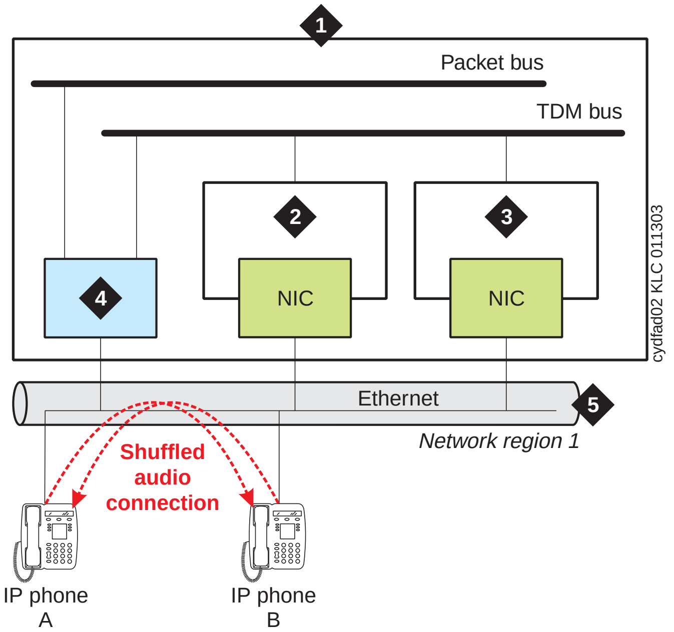


Figure notes:

1. **Avaya server**
2. **TN2302AP IP Media Processor circuit pack**
3. **TN2302AP IP Media Processor circuit pack**
4. **TN799 Control LAN (C-LAN) circuit pack**
5. **LAN/WAN segment administered in Communication Manager as network region 1.**

Figure 4: Shuffled audio connection between IP endpoints in the same network region on page 87 is a schematic of a shuffled connection between two IP endpoints within the same network region. After the call is shuffled, the IP Media Processors are out of the audio connection, and those channels are free to serve other media connections.

Shuffling between different network regions

Figure 5: Shuffled audio connection between IP endpoints in different network regions

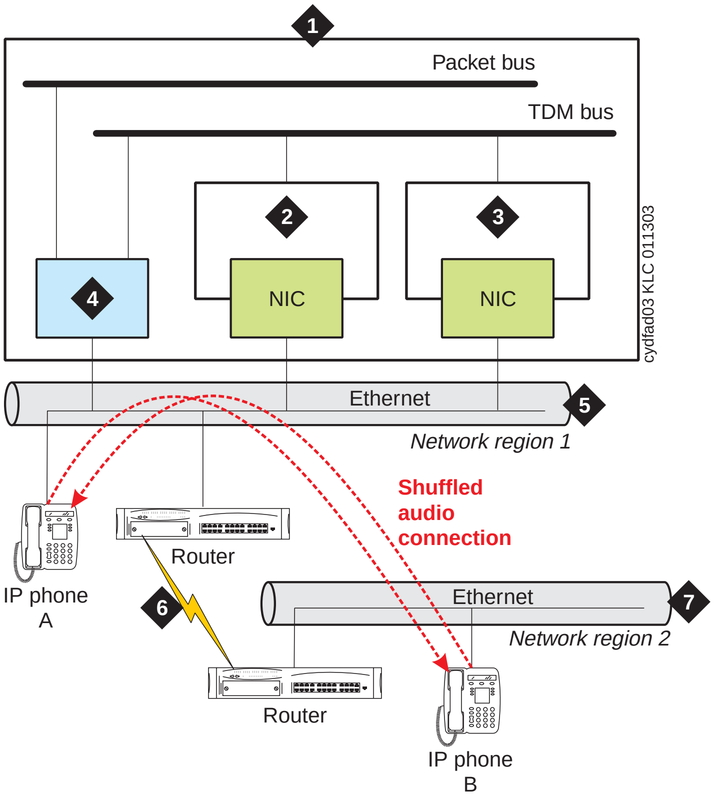


Figure notes:

1. **Avaya server**
2. **TN2302AP IP Media Processor circuit pack**
3. **TN2302AP IP Media Processor circuit pack**
4. **TN799 Control LAN (C-LAN) circuit pack**
5. **LAN/WAN segment administered in Communication Manager as network region 1.**
6. **IP voice packet path between LAN routers**
7. **LAN/WAN segment administered in Communication Manager as network region 2.**

Figure 5: Shuffled audio connection between IP endpoints in different network regions on page 88 is a schematic of a shuffled audio connection between two IP endpoints that are in different network regions that are interconnected and the inter-network region connection management rules are met. After the call is shuffled, both Media Processors are bypassed, making those resources available to serve other media connections. The voice packets from IP endpoints flow directly between LAN routers.

Determining whether an endpoint supports shuffling

Placing a test call from an endpoint that is capable of shuffling to another endpoint whose shuffling capability is unknown can help you to determine whether an endpoint supports audio shuffling or not.

To determine whether an endpoint supports shuffling:

1. Administer the **Direct IP-IP Audio Connections** field on page 2 as **y** (yes) on both endpoint's station form (**`change station`** ***`extension`***).
2. From the endpoint that can support shuffling, place a call to the endpoint that you are testing.

   Wait 2 minutes.
3. At the SAT type **`status station`** ***`extension`*** (administered extension of the endpoint that you are testing) and press **Enter** to display the **Station** screen for this extension.
4. Note the **Port** field value in the **GENERAL STATUS** section of page 1.
5. Scroll to page 4

   In the **AUDIO CHANNEL** section note the value of the **Audio** field under the **Switch Port** column.

   - If the values are the same, the endpoint is capable of shuffling.

     Administer the **Direct IP-IP Audio Connections** field (**`change station`** ***`extension`***, page 2) as **y** (yes).
   - If the values are different, then the endpoint cannot shuffle calls.

     Administer the **Direct IP-IP Audio Connections** field (**`change station`** ***`extension`***, page 2) as **n** (no).

Administrable loss plan

To prevent audio levels from changing when a 2-party call changes from the TDM bus to a shuffled or hairpinned connection, two party connections between IP endpoints are not subject to the switch's administrable loss plan. Although IP endpoints can be assigned to administrable loss groups, the switch is only able to change loss on IP Softphone calls including circuit-switched endpoints. Conference calls of three parties or more are subject to the administrable loss plan, whether those calls involve IP endpoints or not.

What are hairpinned audio connections

Hairpinning means rerouting the voice channel connecting two IP endpoints so that the voice channel goes through the TN2302AP IP Media Processor circuit pack in IP format instead of through the TDM bus. Communication Manager provides only shallow hairpinning, meaning that only the IP and Real Time Protocol (RTP) packet headers are changed as the voice packets go through the TN2302AP circuit pack. This requires that both endpoints use the same codec (coder/decoder), a circuit that takes a varying-voltage analog signal through a digital conversion algorithm to its digital equivalent or vice-versa (digital to analog). Throughout this section, when the word "hairpin" is used, it means shallow hairpinning.

What are the criteria for hairpinning

Communication Manager uses the following criteria to determine whether to hairpin the connection:

- A point-to-point voice connection exists between two endpoints.
- The endpoints are in the same network region, or in different, interconnected regions.
- A single TN2302AP IP Media Processor circuit pack serves both endpoints.
- The endpoints use a single, common codec.
- The endpoints are administered for hairpinning: the **Direct IP-IP Audio Connections** field on the Station screen on page 104 or the Signaling group screen on page 103) is **y.**
- If the **IP Audio Hairpinning** field is **y** (yes), but during registration the endpoint indicates that it does not hairpinning, then a call cannot be hairpinned.

  If the **IP Audio Hairpinning** field is **n** (no), but during registration the endpoint indicates that it can support hairpinning, then calls to that endpoint cannot be hairpinned, giving precedence to the endpoint administration.
- The Determining if shuffling is possible on page 86 are *not* met.
- Both endpoints can connect through the same TN2302AP IP Media Processor circuit pack.

What is an example hairpinned call

Hairpinned audio connections:

- Set up within approximately 50 ms
- Preserve the Real-Time Protocol (RTP) header (for example the timestamp and packet sequence number).
- Do not require volume adjustments on Avaya endpoints, however non-Avaya endpoints might require volume adjustment after the hairpinned connection is established.

Figure 6: Hairpinned audio connection between 2 IP endpoints in the same network region on page 91 is a schematic of a hairpinned audio connection between two IP endpoints in the same network region.

Figure 6: Hairpinned audio connection between 2 IP endpoints in the same network region

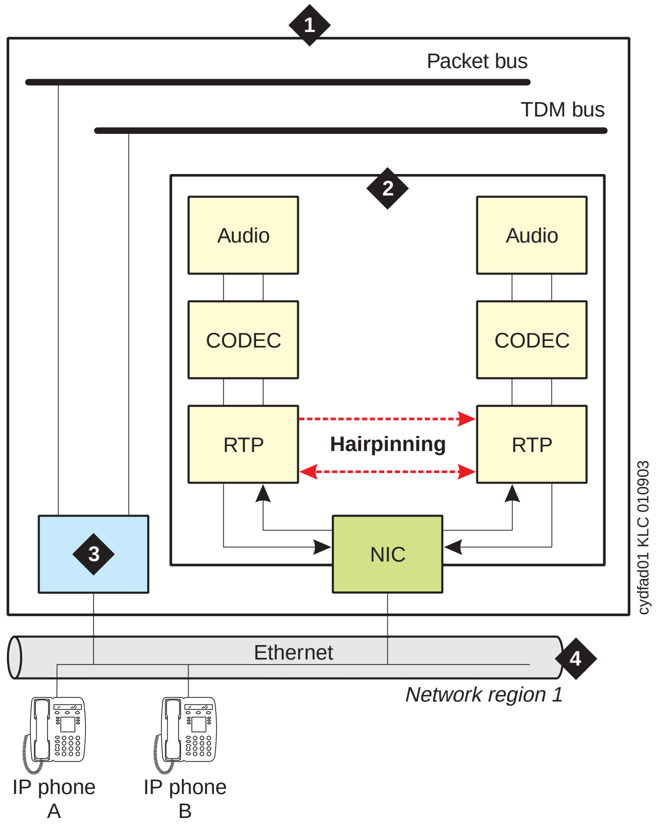


Figure notes:

1. **Avaya server**
2. **TN2302AP IP Media Processor circuit pack**
3. **TN799 Control LAN (C-LAN) circuit pack**
4. **LAN/WAN segment administered in Communication Manager as network region 1.**

Figure 6: Hairpinned audio connection between 2 IP endpoints in the same network region on page 91 shows that hairpinned calls bypass the TN2302AP's codec, thus freeing those resources for other calls. The necessary analog/digital conversions occur in the common codec in each endpoint.

What causes a hairpinned call to be redirected

Whenever a third party is conferenced into a hairpinned call or a tone or announcement must be inserted into the connection, the hairpinned connection is broken and the call is re-routed over the TDM bus.

Determining which TN2302AP circuit pack is hairpinning

Whenever a TN2302AP IP Media Processor circuit pack is hairpinning any calls, its yellow LED is on steady. Although there is no simple way to identify all of the extension numbers that are hairpinning through a particular TN2302AP circuit pack, you can determine which TN2302AP circuit pack a particular extension is using for hairpinning.

To determine which TN2302AP circuit pack is hairpinning:

1. At the SAT, type **`status station`** ***`extension`*** and press **Enter** to display the **Station** form for that extension.
2. Scroll to page 4 of the report.
3. In the **AUDIO CHANNEL** section, check whether there is a value in the **Audio** field under the **Switch Port** column.

   If there is no port listed, then the call is hairpinned.

Hairpinning and shuffling administration interdependencies

Table 10: Hairpinning and shuffling administration on page 92 summarizes the Communication Manager interdependencies that enable hairpinning and shuffling audio connections.

Note:

In order to use hairpinning or shuffling with either Category A or B features, the **Software Version** field (**`list configuration software-versions`**) must be **R9** or greater.

Table 10: Hairpinning and shuffling administration ***1 of 2***

Administration screen	Required customer options[1]	Other interactions
Station	IP Stations Remote Office	Hairpinning is not available if **Service Link Mode** field on ***Station*** form is **permanent**. Shuffling is available only for these endpoints[2]: ● Avaya IP telephone R2 ● Avaya IP Softphone (R2 or older)
Signaling group	H.323 Trunks	

1 of 2

Table 10: Hairpinning and shuffling administration ***2 of 2***

Administration screen	Required customer options[1]	Other interactions
Inter network region	H.323 Trunks IP Stations Remote Office	User login must have features permissions.
Feature-Related System Parameters	H.323 Trunks IP Stations Remote Office	

2 of 2

1. The fields listed in this column must be enabled through the License File. To determine whether these customer options are enabled, use the **`display system-parameters customer-options`** command. If any of the fields listed in this column are not enabled, then either the fields for hairpinning and shuffling are not displayed or, in the case of the **Inter Network Region Connection Management** form, the second page (the actual region-to-region connection administration) does not display.
2. Although other vendors' fully H.323v2-compliant products should have shuffling capability, you should test that before administering such endpoints for hairpinning or shuffling. See the section titled Determining whether an endpoint supports shuffling on page 89.

About Network Address Translation (NAT)

Network address translation (NAT) is a function, typically in a router or firewall, by which an internal IP address is translated to an external IP address. The terms "internal" and "external" are generic and ambiguous, and they are more specifically defined by the application. For example, the most common NAT application is to facilitate communication from hosts on private networks to hosts on the public Internet. In such a case, the internal addresses are private addresses, and the external addresses are public addresses.

Note:

This common NAT application does not use a web proxy server, which would be an entirely different scenario.

Another common NAT application is for some VPN clients. The internal address in this case is the physical address, and the external address is the virtual address. This physical address does not necessarily have to be a private address as shown here, as the subscriber could pay for a public address from the broadband service provider. But regardless of the nature of the physical address, the point is that it cannot be used to communicate back to the enterprise through a VPN tunnel. Once the tunnel is established, the enterprise VPN gateway assigns a virtual address to the VPN client application on the enterprise host. This virtual address is part of the enterprise IP address space, and it must be used to communicate back to the enterprise.

The application of the virtual address varies among VPN clients. Some VPN clients integrate with the operating system in such a way that packets from IP applications (for example, FTP or telnet) on the enterprise host are sourced from the virtual IP address. That is, the IP applications inherently use the virtual IP address. With other VPN clients this does not occur. Instead, the IP applications on the enterprise host inherently use the physical IP address, and the VPN client performs a NAT to the virtual IP address. This NAT is no different than if a router or firewall had done the translation.

What are the types of NAT

Static 1-to-1 NAT

Static 1-to-1 NAT is what has already been covered up to this point. In static 1-to-1 NAT, for every internal address there is an external address, with a static 1-to-1 mapping between internal and external addresses. It is the simplest yet least efficient type of NAT, in terms of address preservation, because every internal host requires an external IP address. This limitation is often impractical when the external addresses are public IP addresses. Sometimes the primary reason for using NAT is to preserve public IP addresses, and for this case there are two other types of NAT: many-to-1 and many-to-a-pool.

Dynamic Many-to-1 NAT

Dynamic many-to-1 NAT is as the name implies. Many internal addresses are dynamically translated to a single external address. Multiple internal addresses can be translated to the same external address, when the TCP/UDP ports are translated in addition to the IP addresses. This is known as network address port translation (NAPT) or simply port address translation (PAT). It appears to the external server that multiple requests are coming from a single IP address, but from different TCP/UDP ports. The NAT device remembers which internal source ports were translated to which external source ports.

In the simplest form of many-to-1 NAT, the internal host must initiate the communication to the external host, which then generates a port mapping within the NAT device, allowing the external host to reply back to the internal host. It is a paradox with this type of NAT (in its simplest form) that the external host cannot generate a port mapping to initiate the communication with the internal host, and without initiating the communication, there is no way to generate the port mapping. This condition does not exist with 1-to-1 NAT, as there is no mapping of ports.

Dynamic Many-to-a-Pool NAT

Many-to-a-pool NAT combines some of the characteristics of both 1-to-1 and many-to-1 NAT. The general idea behind many-to-a-pool NAT is that a 1-to-1 mapping is not desired, but there are too many internal hosts to use a single external address. Therefore, a pool of multiple external addresses is used for NAT. There are enough external addresses in the pool to support all the internal hosts, but not nearly as many pool addresses as there are internal hosts.

What are the issues between NAT and H.323

Some of the hurdles that NAT presents to H.323 include:

- H.323 messages, which are part of the IP payload, have embedded IP addresses in them.

  NAT translates the IP address in the IP header, but not the embedded addresses in the H.323 messages. This is a problem that can be and has been addressed with H.323-aware NAT devices. It has also been addressed with Avaya Communication Manager 1.3 and later versions of the NAT feature.

- When an endpoint (IP telephone) registers with the gatekeeper (call server), that endpoint's IP address must stay the same for the duration of the registration.

  This rules out almost all current implementations of many-to-a-pool NAT.

- TCP/UDP ports are involved in all aspects of IP telephony — endpoint registration, call signaling, and RTP audio transmission.

  These ports must remain unchanged for the duration of an event, duration of the registration, or duration of a call. Also, the gatekeeper must know ahead of time which ports will be used by the endpoints for audio transmission, and the ports can vary per call. These requirements make it very difficult for H.323 to work with NAPT, which rules out almost all current implementations of many-to-1 and many-to-a-pool NAT.

Avaya Communication Manager NAT Shuffling feature

The Avaya Communication Manager NAT Shuffling feature permits IP telephones and IP Softphones to work behind a NAT device. This feature was available prior to release 1.3, but it did not work with shuffled calls (**Direct IP-IP Audio** enabled). The NAT feature now works with shuffled calls.

Terms:

The following terms are used to describe the NAT Shuffling feature:

- Native Address — The original IP address configured on the device itself (internal address)
- Translated Address — The IP address after it has gone through NAT, as seen by devices on the other side of the translation (external address)

- Gatekeeper — The Avaya device that is handling call signaling.

  It could be a portal to the gatekeeper, such as a C-LAN, or the gatekeeper itself, such as an S8300 Media Server.
- Gateway — The Avaya device that is handling media conversion between TDM and IP, such as a MedPro board, G700 VoIP Media Module, or G350 Media Gateway.

The essence of this feature is that Communication Manager keeps track of the native and translated IP addresses for every IP station (IP telephone or IP Softphone). If an IP station registration appears with different addresses in the IP header and the RAS message, the call server stores the two addresses and alerts the station that NAT has taken place.

This feature works with static 1-to-1 NAT. It does not work with NAPT, so the TCP/UDP ports sourced by the IP stations must not be changed. Consequently, this feature does not work with many-to-1 NAT. This feature *may* work with many-to-a-pool NAT, if a station's translated address remains constant for as long as the station is registered, and there is no port translation.

The NAT device must perform plain NAT – not H.323-aware NAT. Any H.323-aware feature in the NAT device must be disabled, so that there are not two independent devices trying to compensate for H.323 at the same time.

Rules:

The following rules govern the NAT Shuffling feature. The **Direct IP-IP Audio** parameters are configured on the SAT **ip-network-region** form.

1. When **Direct IP-IP Audio** is enabled (default) and a station with NAT and a station without NAT talk to one another, the translated address is always used.
2. When two stations with NAT talk to one another, the native addresses are used (default) when **Yes** or **Native (NAT)** is specified for **Direct IP-IP Audio**, and the translated addresses are used when **Translated (NAT)** is specified.
3. The Gatekeeper and Gateway must *not* be enabled for NAT. As long as this is true, they may be assigned to any network region.

Administering hairpinning and shuffling

Choosing how to administer hairpinning and shuffling

You can administer shuffled and hairpinned connections:

- Independently for system-wide applicability
- Within a network region
- At the user level

Table 11: Hairpinning and shuffling administration on page 97 lists the forms and provides links to all three levels:

Table 11: Hairpinning and shuffling administration

Level	Communication Manager form	Link to procedure
System	Feature-Related System Parameters	Administering hairpinning and shuffling at the system-level on page 97
Network region	Network Region	Administering hairpinning and shuffling in network regions on page 99
User		
IP Trunks	Signaling Group	Administering H.323 trunks for hairpinning and shuffling (S8500/S8700) on page 102
IP endpoints	Station	Administering IP endpoints for hairpinning and shuffling on page 104

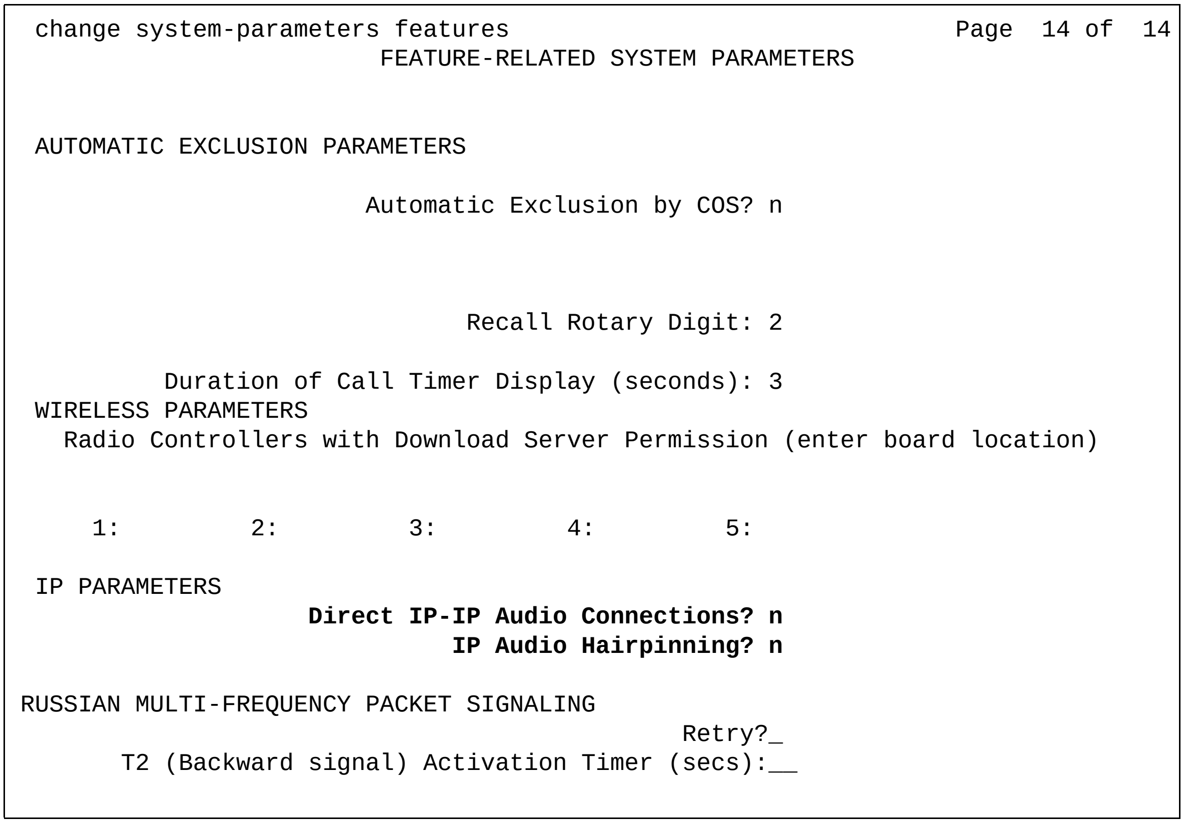
Administering hairpinning and shuffling at the system-level

You can administer hairpinning or shuffling as a system-wide parameter.

To administer hairpinning and shuffling as a system-level parameter

1. At the SAT, type **`change system-parameters features`** and press **Enter** to display the **Feature-Related System Parameters** screen:

Feature-Related System Parameters screen



```
 change system-parameters features                                 Page  14 of  14
                         FEATURE-RELATED SYSTEM PARAMETERS


 AUTOMATIC EXCLUSION PARAMETERS

                        Automatic Exclusion by COS? n



                               Recall Rotary Digit: 2

          Duration of Call Timer Display (seconds): 3
 WIRELESS PARAMETERS
   Radio Controllers with Download Server Permission (enter board location)


     1:          2:          3:          4:          5:

 IP PARAMETERS
                      Direct IP-IP Audio Connections? n
                                IP Audio Hairpinning? n

RUSSIAN MULTI-FREQUENCY PACKET SIGNALING
                                              Retry?_
       T2 (Backward signal) Activation Timer (secs):__
```

2. To allow shuffled IP calls using a public IP address (default), set the **Direct IP-IP Audio Connections** field to **y**.

   To disallow shuffled IP calls set this field to **n**. Be sure that you understand the interactions in Hairpinning and shuffling administration interdependencies on page 92 and the notes below.

3. To allow hairpinned audio connections, type **y** (yes) in the **IP Audio Hairpinning** field, noting the interactions in Hairpinning and shuffling administration interdependencies on page 92 and the notes below.

4. Save the changes.

   Note:

   > The **Direct IP-IP Audio Connections** and **IP Audio Hairpinning** fields do not display if the **IP Stations** field, the **H.323 Trunks** field, and the **Remote Office** field on the **Customer Options** form are set to **n**.

Administering hairpinning and shuffling in network regions

Inter-network region connection management

Shuffling and hairpinning endpoints or media processing resources in any given network region is independently administered per network region, which uses a matrix to define the desired connections between pairs of regions.

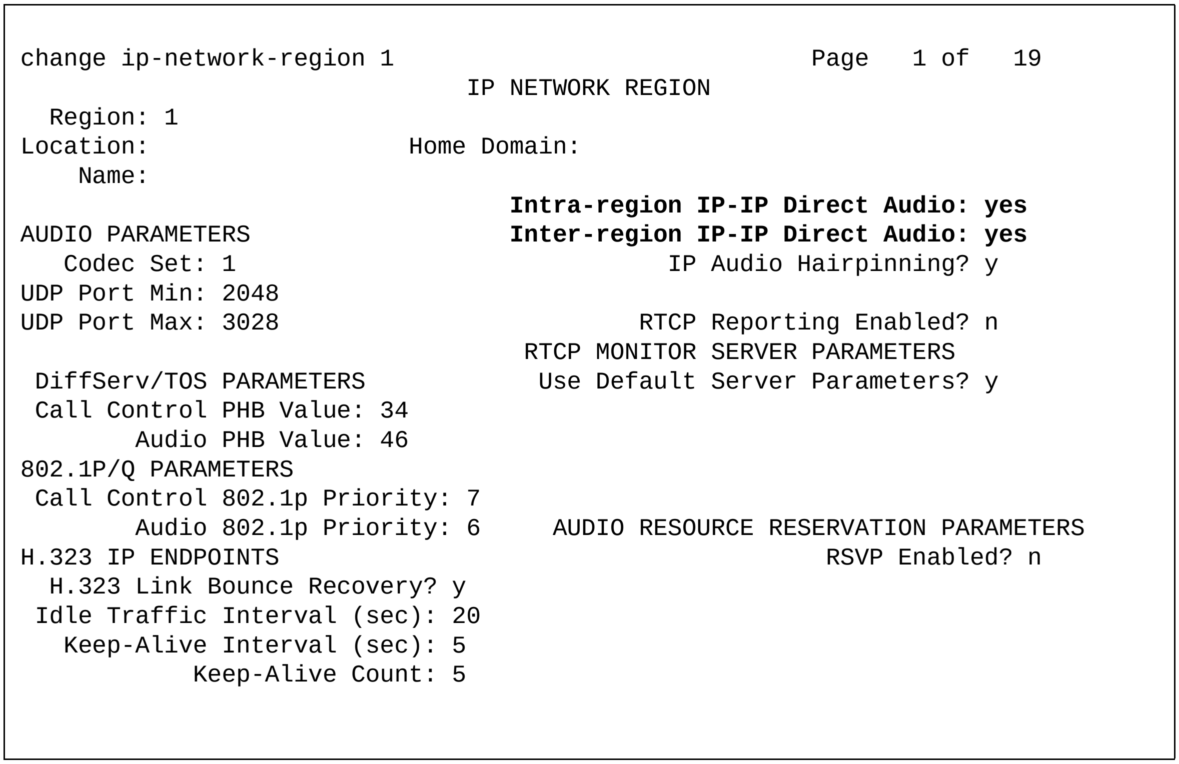
The matrix is used two ways:

- It specifies what regions are valid for resource allocation when resources in the preferred region are unavailable.
- When a call exists between two IP endpoints in different regions, the matrix specifies whether those two regions can be directly connected.

To administer hairpinning or shuffling within a network region:

1. At the SAT type **`change ip-network-region`** ***`number`*** and press **Enter** to display the **IP Network Region** screen.

IP Network Region screen



```
change ip-network-region 1                             Page   1 of   19
                               IP NETWORK REGION
  Region: 1
Location:                  Home Domain:
    Name:
                                  Intra-region IP-IP Direct Audio: yes
AUDIO PARAMETERS                  Inter-region IP-IP Direct Audio: yes
   Codec Set: 1                              IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 3028                         RTCP Reporting Enabled? n
                                   RTCP MONITOR SERVER PARAMETERS
 DiffServ/TOS PARAMETERS            Use Default Server Parameters? y
 Call Control PHB Value: 34
        Audio PHB Value: 46
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6     AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

2. Administer the **IP-IP Direct Audio** fields:

   - The **Intra-region IP-IP Direct Audio** field permits shuffling if both endpoints are in the same region.
   - The **Inter-region IP-IP Direct Audio** field permits shuffling if the two endpoints are in two different regions.

   The allowable values for both fields are:

   - **y** -- permits shuffling the call
   - **n** -- disallows shuffling the call
   - **native**-- the IP address of a phone itself, or no translation by a Network Address Translation (NAT) device
   - **translated** -- the translated IP address that a Network Address Translation (NAT) device provides for the native address

   Note:

   > If there is no NAT device in use at all, then the native and translated addresses are the same. For more information on NAT, see the *Administrator's Guide for Avaya Communication Manager, 555-233-506* and *Avaya Application Solutions IP Telephony Deployment Guide,* 555-245-600*.*

   Note:

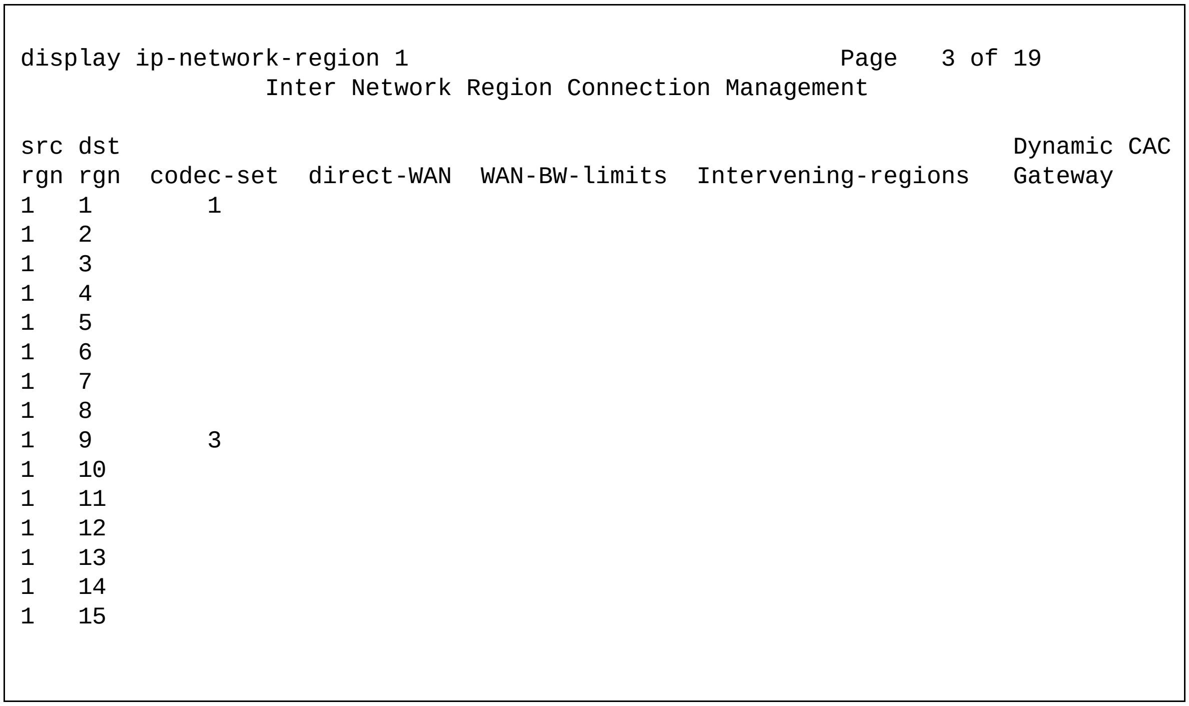
   > The hairpinning and shuffling fields on the **IP Network Regions** screen do not display unless the **IP Stations**, the **H.323 Trunks**, or the **Remote Office** field is set to **y** (yes) on the **Optional Features** (**`display system-parameter customer-options`**) screen. These features must be enabled in the system's License File.

3. Go to page 3 and administer the common codec sets on the **Inter Network Region Connection Management** form (Inter Network Region Connection Management form on page 101).

   Note:

   > You cannot connect IP endpoints in different network regions or share TN799 C-LAN or TN2032 IP Media Processor resources between/among network regions unless you make a codec entry in this matrix specifying the codec set to be used.

Inter Network Region Connection Management form



```
display ip-network-region 1                              Page    3 of 19
                 Inter Network Region Connection Management

src dst                                                              Dynamic CAC
rgn rgn   codec-set   direct-WAN  WAN-BW-limits  Intervening-regions   Gateway
1   1        1
1   2
1   3
1   4
1   5
1   6
1   7
1   8
1   9        3
1   10
1   11
1   12
1   13
1   14
1   15
```

Fro example, in Inter Network Region Connection Management form on page 101, network region 1 communicates with:

- Network region 1 using codec set 1
- Network region 9 using codec set 3

Note:

Use the **`list ip-codec-set`** command for a list of codecs.

4. Save the changes.

Administering and selecting codecs

When an IP endpoint calls another IP endpoint, Communication Manager asks that the 2nd endpoint choose the same codec that the 1st endpoint offered at call setup. However, if the 2nd endpoint cannot match codecs with the 1st's, the call is set up with each endpoint's administered (preferred) codec, and the data streams are converted between them, often resulting in degraded audio quality because of the series of different compressions/ decompressions or multiple use of the same codec.

When an endpoint (station or trunk) initially connects to the server, Communication Manager selects the first codec that is common to both the server and the endpoint. The **Inter Network Region Connection Management** screen specifies codec set(s) to use *within* an individual region (intra-region) and a codec set to use *between/among* (inter-region) network regions. Depending upon the network region of the requesting H.323 endpoint or trunk and the network region of the TN2302AP IP Media Processor circuit pack:

- If the endpoint and the TN2302AP are in same region, the administered intra-region codec set is chosen.
- If the endpoint and the TN2302AP are in different regions, the administered inter-region codec set is chosen.

For example, a region might have its intra-network codec administered as G.711 as the first choice, followed by the other low bit rate codecs. The **Inter Network Region Connection Management** screen for the inter-network region might have G.729 (a low-bit codec that preserves bandwidth) as the only choice. Initially, when a call is set up between these two interconnected regions, the TN2302AP IP Media Processor provides the audio stream conversion between G.711 and G.729. When the media stream is shuffled away from a TDM-based connection, the two endpoints can use only the G.729 codec.

Note:

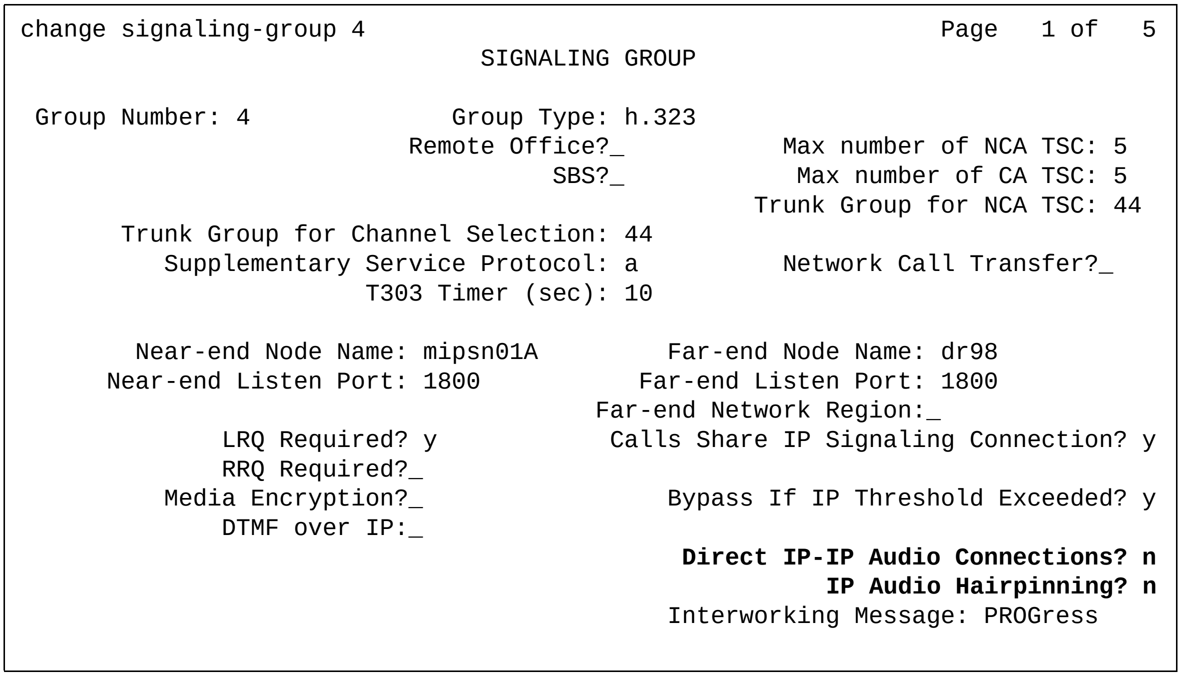
> If you are administering an H.323 trunk that uses Teletype for the Deaf (TTD), use the G.711 codec as the primary choice for those trunks. This ensures accurate TTD tone transmission through the connection.

Administering H.323 trunks for hairpinning and shuffling (S8500/S8700)

To administer an H.323 trunk for hairpinning or shuffling

1. At the SAT, type **`change signaling group`** ***`number`*** and press **Enter** to display the **Signaling Group** screen (Signaling group screen on page 103).

Signaling group screen



```
change signaling-group 4                                         Page   1 of   5
                                SIGNALING GROUP

 Group Number: 4             Group Type: h.323
                          Remote Office?_          Max number of NCA TSC: 5
                                    SBS?_           Max number of CA TSC: 5
                                                  Trunk Group for NCA TSC: 44
       Trunk Group for Channel Selection: 44
          Supplementary Service Protocol: a        Network Call Transfer?_
                        T303 Timer (sec): 10

        Near-end Node Name: mipsn01A         Far-end Node Name: dr98
      Near-end Listen Port: 1800           Far-end Listen Port: 1800
                                        Far-end Network Region:_
              LRQ Required? y            Calls Share IP Signaling Connection? y
              RRQ Required?_
          Media Encryption?_                 Bypass If IP Threshold Exceeded? y
              DTMF over IP:_
                                              Direct IP-IP Audio Connections? n
                                                        IP Audio Hairpinning? n
                                             Interworking Message: PROGress
```

2. To allow shuffled IP calls using a public IP address (default), set the **Direct IP-IP Audio Connections** field to **y**.

   To disallow shuffled IP calls set this field to **n**. Be sure that you understand the interactions in Hairpinning and shuffling administration interdependencies on page 92 and the notes below.

3. To allow hairpinned audio connections, type **y** (yes) in the **IP Audio Hairpinning** field, noting the interactions in Hairpinning and shuffling administration interdependencies on page 92 and the notes below.

4. Save the changes.

Note:

The hairpinning and shuffling fields on the **Signaling Group** screen do not display unless either the **H.323 Trunks** or **Remote Office** field is set to **y** (yes) on the **Optional Features** (**`display system-parameters customer-options`**) screen. These features must be enabled in the system's License File.

Note:

If you are administering an H.323 trunk that uses Teletype for the Deaf (TTD), use the G.711 codecs as the primary codec choice for those trunks to ensure accurate TTD tone transmission through the connection.

Administering IP endpoints for hairpinning and shuffling

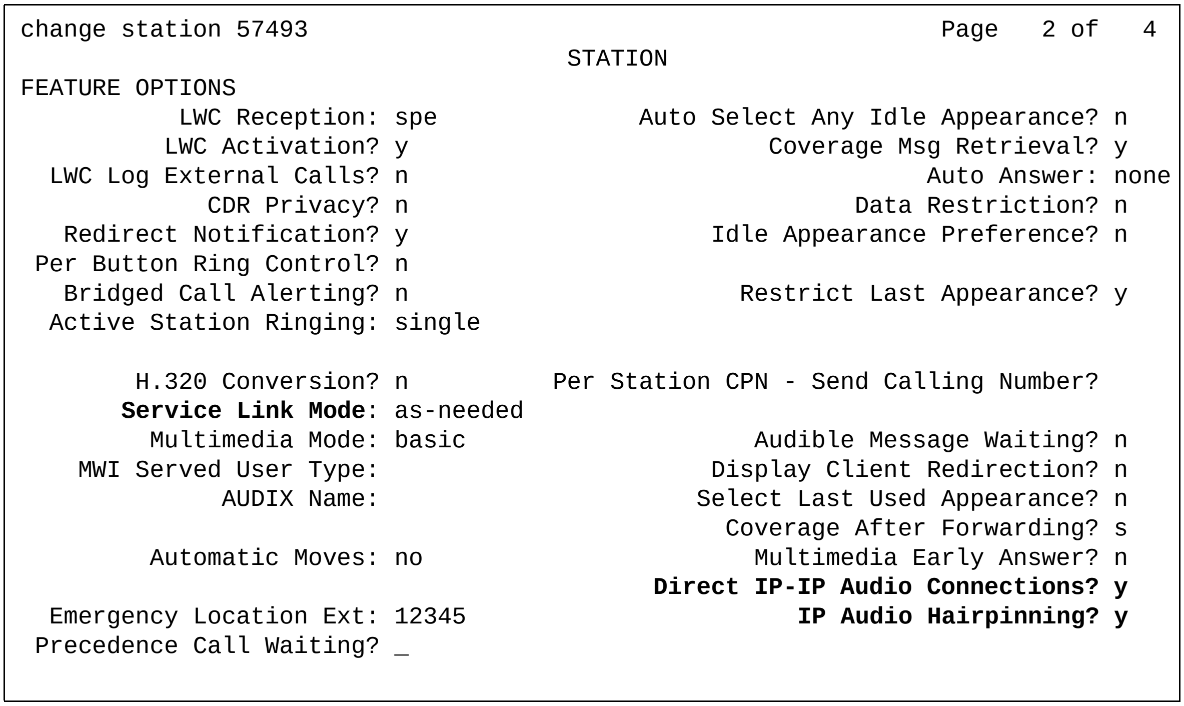
Whether any given station is allowed to shuffle or hairpin is independently administered per endpoint on the **Station** screen. The specific station types that you can administer for hairpinning or shuffling are:

- All Avaya IP stations
- Other vendors' H.323-compatible stations

To administer an IP endpoint for hairpinning or shuffling

1. At the SAT, type **`change station`** ***`extension`*** and press **Enter** to display the **Station** screen (Station screen on page 104)

Station screen



```
change station 57493                                              Page   2 of   4
                                      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
   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                      Audible Message Waiting? n
     MWI Served User Type:                        Display Client Redirection? n
              AUDIX Name:                        Select Last Used Appearance? n
                                                   Coverage After Forwarding? s
         Automatic Moves: no                         Multimedia Early Answer? n
                                             Direct IP-IP Audio Connections? y
  Emergency Location Ext: 12345                          IP Audio Hairpinning? y
 Precedence Call Waiting? _
```

2. To allow shuffled IP calls using a public IP address (default), set the **Direct IP-IP Audio Connections** field to **y**.

   To disallow shuffled IP calls set this field to **n**. Be sure that you understand the interactions in Hairpinning and shuffling administration interdependencies on page 92 and the notes below.

3. To allow hairpinned audio connections, type **y** in the **IP Audio Hairpinning** field, noting the interactions in Hairpinning and shuffling administration interdependencies on page 92 and the notes below.

4. Save the changes.

Note:

The hairpinning and shuffling fields on the **Station** screen do not display unless either the **IP Stations** or **Remote Office** field is set to **y** (yes) on the **Optional Features** (`display system-parameter customer-options`) screen. These features must be enabled in the system's License File.

Note:

The **Direct IP-IP Audio Connections** field cannot be set to **y** if the **Service Link Mode** field is set to **permanent**.

Contradictory IP station administration

- If an IP station is administered for dual-connect, and if the two extension numbers for that station have differing values administered in their **Direct IP-IP audio Connections** fields, then the station cannot shuffle calls.
- If an IP station is administered for dual-connect, and if the two extension numbers for that station have differing values administered in their **IP-IP Audio Hairpinning** fields, then the station cannot hairpin calls.

IP stations used for call center service-observing

If a Call Center agent is active on a shuffled call, and a Call Center supervisor wants to service-observe the call, the agent might notice the 200 ms break in the speech while the call is redirected to the TDM bus. For this reason, Avaya recommends that you administer the shuffling and hairpinning fields as **n** (no) for stations that are used for service-observing.

Upgrade interactions for hairpinning and shuffling

Upgrading from Release 8 to Release 9 or higher

If the **Inter-region IP connectivity allowed** field on the IP-interfaces form is **y** (yes), then the entries for all interconnections among regions 1 to 10 on the new (upgraded) **IP-Network Region** form are set to **1**.

Upgrading to Release 9 or higher

The **Direct IP-IP Audio Connections** and **IP Audio Hairpinning** fields default to **n** (no), just as they do for a new installation.

Administering IP endpoint signal loss

The amount of loss applied between any two endpoints on a call is administrable. However, the Telecommunications Industry Association (TIA) has published standards for the levels that IP endpoints should use. The IP endpoints will always transmit audio at TIA standard levels, and expect to receive audio at TIA standard levels. If an IP audio signal goes to or comes from the TDM bus through a TN2302AP Media Processor, the circuit pack will adjust the levels to be approximately equal the levels of a signal to or from a Digital Communications Protocol (DCP) set. By default, IP endpoints are the same loss group as DCP sets, Group 2.

Adjusting loss to USA DCP levels

The switch instructs the TN2302AP Media Processor circuit pack to insert loss into the signal coming from the IP phone, and insert gain in the signal going to the IP phone, to equal the levels of a signal to or from a DCP set.

Note:

The voice level on a shuffled call is not affected by entries administered in the **2-Party Loss Plan** screen.

Note:

The loss that is applied to a hairpinned or shuffled audio connection is constant for all three connection types:

- station-to-station
- station-to-trunk
- trunk-to-trunk

Administering FAX, Modem, and TTY calls over IP Trunks

Avaya Communication Manager transports fax, modem, and TTY calls over a VoIP network using relay mode (see What relay mode is on page 107), pass through mode (see What pass through mode is on page 108), or both. As a result, Communication Manager supports transport of the following:

- Teletypewriter device (TTY) tone relay over the corporate IP intranet and the Internet
- Faxes over a corporate IP intranet

  Note:

  The path between endpoints for fax transmissions must use Avaya telecommunications and networking equipment.

  Note:

  Faxes sent to non-Avaya endpoints cannot be encrypted.

- T.38 Fax over the Internet (including endpoints connected to non-Avaya systems)
- Modem tones over a corporate IP intranet

  The path between endpoints for modem tone transmissions must use Avaya telecommunications and networking equipment.

What relay mode is

In relay mode, the firmware on the device (the G700/G350 media gateway, the MM760 VoIP media module or TN2302 Media Processor) detects the tones of the call (fax, modem, or TTY) and uses the appropriate modulation protocol (for fax or modem) or Baudot transport representation (TTY) to terminate or originate the call so that it can be carried over the IP network. The modulation and demodulation for fax and modem calls reduces bandwidth use over the IP network and improves the reliability of transmission. The correct tones are regenerated before final delivery to the endpoint.

Note:

The number of simultaneous calls that a device (gateway, media module, TN2302) can handle is reduced by modulation and demodulation the device must perform for relay mode.

What pass through mode is

In pass through mode, the firmware on the device (the G700/G350 media gateway, the MM760 VoIP media module or TN2302 Media Processor) detects the tones of the call (fax, modem, or TTY) and uses G.711 encoding to carry the call over the IP network. Pass through mode provides higher quality transmission when endpoints in the network are all synchronized to the same clock source. The call is un-encoded before final delivery to the endpoint.

Note:

Though pass through mode increases the bandwidth usage (per channel), it allows the same number of simultaneous fax/modem calls on the device as the number of simultaneous voice calls. For example, on a G700 Media Gateway, pass through allows 64 simultaneous fax/modem calls instead of only 16 with relay.

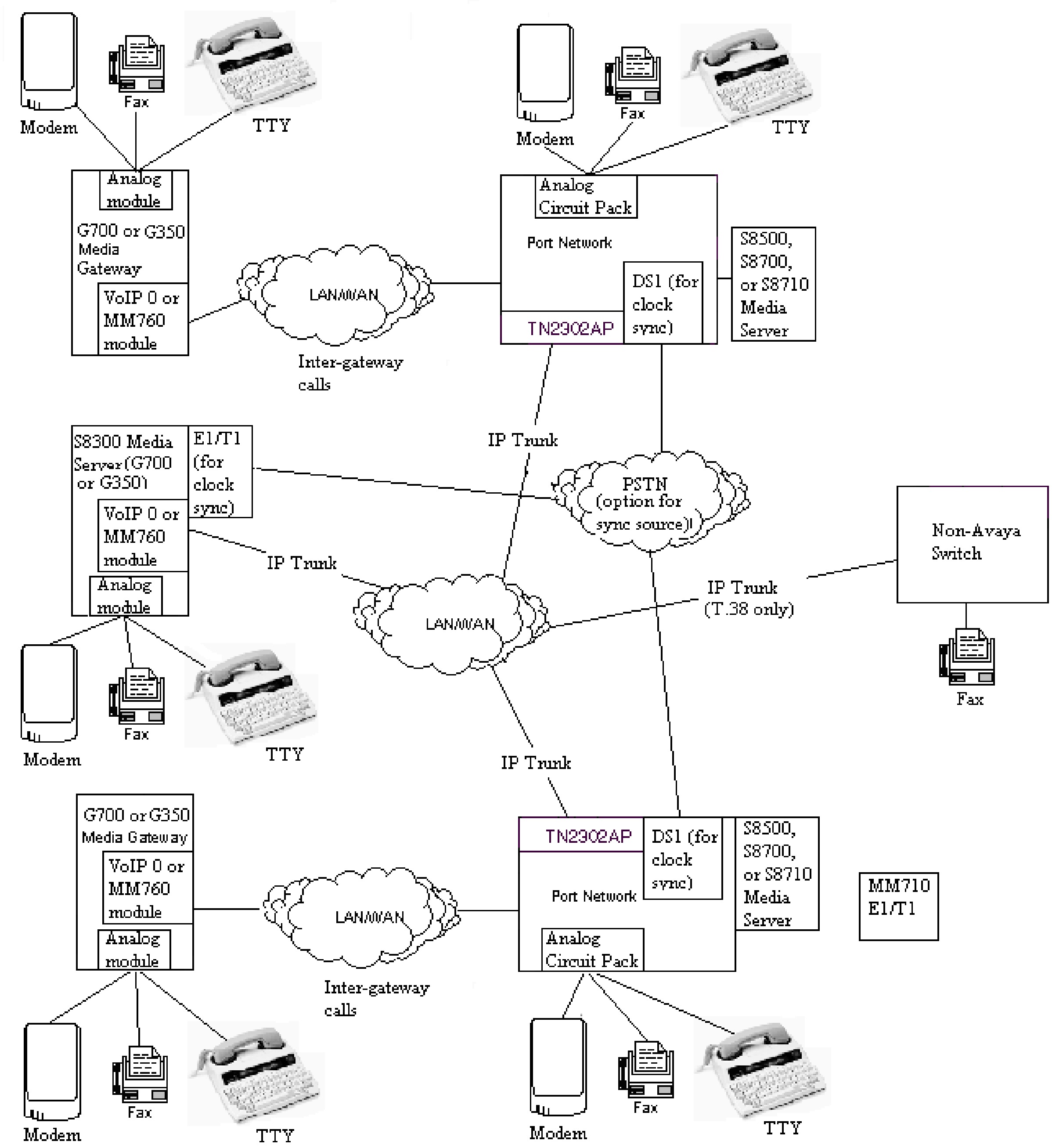
Note:

For pass-through mode on modem and TTY calls over an IP network, the sending and receiving servers should have a common synchronization source. Sychronized clocks can be established by using a source on the public network. See Figure 7: IP network connections over which fax, modem, and TTY calls are made on page 109.

Note:

You cannot send faxes in pass through mode with the T.38 standard.

Figure 7: IP network connections over which fax, modem, and TTY calls are made



Overview of steps to administer fax, TTY, and modem calls over IP trunks

The information in this section assumes the following:

- The endpoints sending and receiving the calls are connected to a private network that uses H.323 trunking or LAN connections between gateways and/or port networks.
- Calls can either be passed over the public network using ISDN-PRI trunks or passed over an H.323 private network to Communication Manager switches that are similarly enabled.

To administer fax, TTY, and modem calls over IP trunks, first consider the following:

- Fax, TTY, and modem transmission modes and speeds on page 111
- Considerations for administering fax, TTY, and modem transmission on page 114
- Bandwidth for fax, modem, and TTY calls over IP networks on page 117
- AES/AEA Media Encryption on page 118

After considering the criteria from the preceding list, complete the following tasks:

1. Create one or more IP Codec sets that enable the appropriate transmission modes for the endpoints on your gateways. See Administering IP CODEC sets on page 127.

   Note:

   You create the fax, modem, and TTY settings (including redundancy) on the second page of the IP Codec Set screen.

2. Assign each codec set to the appropriate network region. See Administering IP network regions on page 133.
3. Assign the network region to the appropriate device(s):
   - TN2302 Media Processor (see Defining IP interfaces on page 67)
   - G350 or G700 Media Gateway (see To add the media gateway on page 61)
4. If the TN2302AP (IP Media Processor) resources are shared between/among administered network regions, administer inter-network region connections. See Interconnecting the network regions on page 139.

Fax, TTY, and modem transmission modes and speeds

Communication Manager provides the following methods for supporting fax, TTY, and modem transmission over IP (see Table 12: Fax, TTY, and modem transmission modes and speeds on page 111).

Table 12: Fax, TTY, and modem transmission modes and speeds ***1 of 3***

Mode	Maximum Rate	Comments
T.38 Fax Standard (relay only)	9600 bps	This capability is standards-based and uses IP trunks and H.323 signaling to allow communication with non-Avaya systems. Additionally, the T.38 fax capability uses the UDP protocol. **Note:** Fax endpoints served by two different Avaya media servers can also send T.38 faxes to each other if both systems are enabled for T.38 fax. In this case, the media servers also use IP trunks. However, if the T.38 fax sending and receiving endpoints are on port networks or media gateways that are registered to the same media server, the gateways or port networks revert to Avaya fax relay mode. Both the sending and receiving systems must announce support of T.38 fax data applications during the H.245 capabilities exchange. Avaya systems announce support of T.38 fax if the capability is administered on the Codec Set screen for the region and a T.38-capable media processor was chosen for the voice channel. In addition, for a successful fax transmission, both systems should support the H.245 null capability exchange (shuffling) in order to avoid multiple IP hops in the connection. **Note:** To use the T.38 fax capability, modem relay and modem pass through must be disabled. Additionally, the T.38 fax capability does not support TCP, fax relay, or fax pass through. You can assign packet redundancy to T.38 standard faxes to improve packet delivery and robustness of fax transport over the network.

1 of 3

Table 12: Fax, TTY, and modem transmission modes and speeds ***2 of 3***

Mode	Maximum Rate	Comments
Fax Relay	9600 bps	Because the data packets for faxes in relay mode are sent almost exclusively in one direction, from the sending endpoint to the receiving endpoint, bandwidth use is reduced.
Fax Pass Through	V.34 (33.6 kbps)	The transport speed is up to the equivalent of circuit-switched calls and supports G3 and Super G3 fax rates. ⚠ **CAUTION:** If users are using Super G3 fax machines as well as modems, do *not* assign these fax machines to a network region with an IP Codec set that is modem-enabled as well as fax-enabled. If its Codec set is enabled for both modem and fax signaling, a Super G3 fax machine incorrectly tries to use the modem transmission instead of the fax transmission. Therefore, assign modem endpoints to a network region that uses a modem-enabled IP Codec set, and assign the Super G3 fax machines to a network region that uses a fax-enabled IP Codec set. You can assign packet redundancy in both pass-through and relay mode, which means the media gateways use packet redundancy to improve packet delivery and robustness of fax transport over the network. Pass through mode uses more network bandwidth than relay mode. Redundancy increases bandwidth usage even more.
TTY Relay	16 kbps	This transport of TTY supports US English TTY (Baudot 45.45) and UK English TTY (Baudot 50). TTY uses RFC 2833 or RFC 2198 style packets to transport TTY characters. Depending on the presence of TTY characters on a call, the transmission toggles between voice mode and TTY mode. The system uses up to 16 kbps of bandwidth, including packet redundancy, when sending TTY characters and normal bandwidth of the audio codec for voice mode.

2 of 3

Table 12: Fax, TTY, and modem transmission modes and speeds ***3 of 3***

Mode	Maximum Rate	Comments
TTY Pass Through	87-110 kbps	In pass-through mode, you can also assign packet redundancy, which means the media gateways send duplicated TTY packets to ensure and improve quality over the network. Pass through mode uses more network bandwidth than relay mode. Pass through TTY uses 87-110 kbps, depending on the packet size, whereas TTY relay uses, at most, the bandwidth of the configured audio codec. Redundancy increases bandwidth usage even more.
Modem Relay	V.32 (9600 bps)	The maximum transmission rate may vary with the version of firmware. The packet size for modem relay is determined by the packet size of the codec selected but is always at least 30ms. Also, each level of packet redundancy, if selected, increases the bandwidth usage linearly (that is, the first level of redundancy doubles the bandwidth usage; the second level of redundancy triples the bandwidth usage, and so on). **Note:** Modem over IP in relay mode is currently available only for use by specific secure analog telephones that meet the Future Narrowband Digital Terminal (FNBDT) standard. See your sales representative for more information. Additionally, modem relay is limited to V.32/ V.32bis data rates.
Modem Pass Through	V.34 (33.6 kbps) and V.90/ V.92 (43.4 kbps)	Transport speed is dependent on the negotiated rate of the modem endpoints. Though the media servers and media gateways support modem signaling at v.34 (33.6 bps) or v.90 and v.92 (43.4 kbps), the modem endpoints may automatically reduce transmission speed to ensure maximum quality of signals. V.90 and V.92 are speeds typically supported by modem endpoints only when directly connected to a service provider Internet service. You can also assign packet redundancy in pass-through mode, which means the media gateways send duplicated modem packets to improve packet delivery and robustness of fax transport over the network. Pass through mode uses more network bandwidth than relay mode. Redundancy increases bandwidth usage even more. The maximum packet size for modem pass through is 20 ms.

3 of 3

Considerations for administering fax, TTY, and modem transmission

There are a number of factors to consider when configuring your system for fax, TTY, and modem calls over an IP network:

- Encryption

  You can encrypt most type of relay and pass through calls using either the Avaya Encryption Algorithm (AEA) or the Advanced Encryption Standard (AES). See AES/AEA Media Encryption on page 118.

- Bandwidth usage

  Bandwidth usage of modem relay varies, depending on packet size used and the redundancy level selected.   The packet size for modem relay is determined by the packet size of the codec selected.   Bandwidth usage of modem pass through varies depending on the redundancy level and packet size selected. The maximum packet size for modem pass through is 20 ms.

  Bandwidth usage for other modes also varies, depending on the packet size used, whether redundant packets are sent, and whether the relay or pass through method is used.

  See Table 13: Bandwidth for fax, modem, and TTY calls over IP networks on page 117 for the bandwidth usage.

- Calls with non-Avaya systems

  For fax calls where one of the communicating endpoints is connected to a non-Avaya communications system, the non-Avaya system and the Avaya system should both have T.38 defined for the associated codecs.

  Modem and TTY calls over the IP network *cannot* be successfully sent to non-Avaya systems.

- Differing transmission methods at the sending/receiving endpoints

  The transmission method or methods used on both the sending and receiving ends of a fax/modem/TTY call should be the same.

  In some cases, a call succeeds even though the transmission method for the sending and receiving endpoints is different. Generally, however, for a call to succeed, the two endpoints must be administered for the same transmission method.

- Hardwarerequirements

  The relay and pass through capabilities require the following hardware:

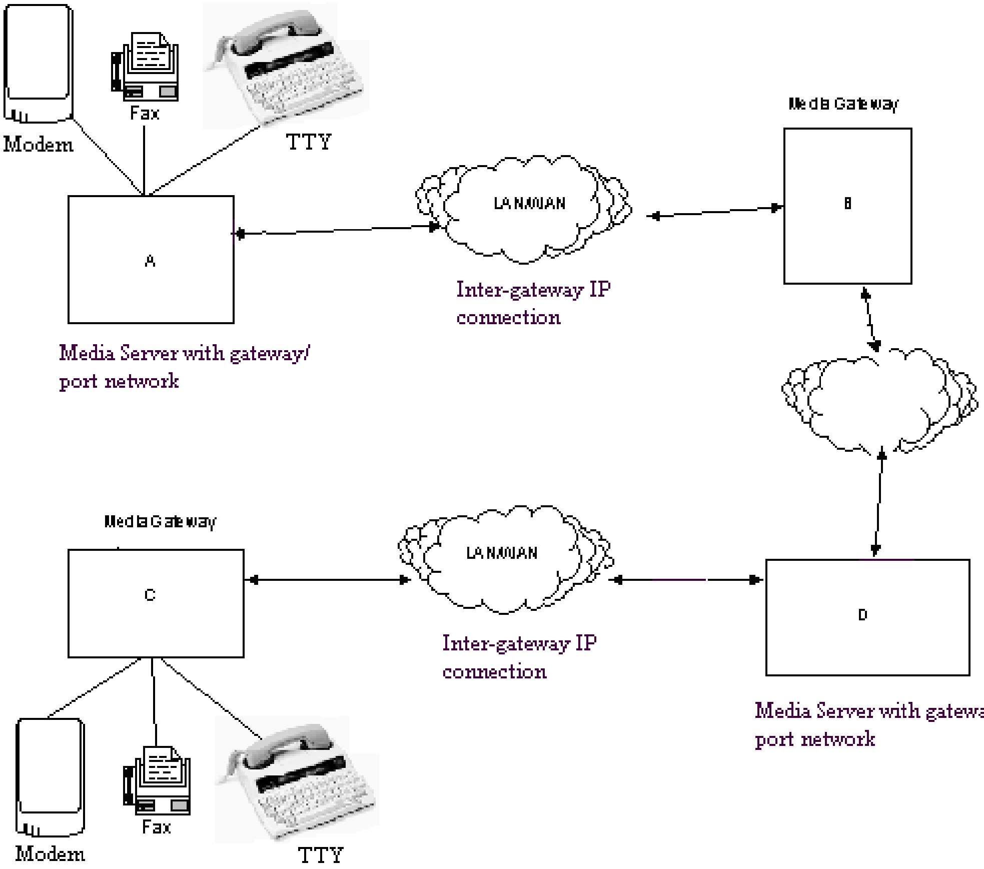
  - For DEFINITY CSI and SI servers, S8500/S8500B Media Servers, or S8700/S8710 Media Servers, Hardware Vintage 10 or later of the TN2302AP circuit pack with firmware version 90 or later is required.
  - For the G700 Media Gateway, G700 firmware version 22.14.0, and VoIP firmware Vintage 40 or greater to support Communication Manager R2.2 is required. An MM760 Media Module with firmware Vintage 40 or greater may be used for additional VoIP capacity.
  - For the G350 Media Gateway, G350 firmware version 22.14.0 or greater to support Communication Manager R2.2 is required.
  - For T.38 fax capability, endpoints on other non-Avaya T.38 compliant communications systems may send fax calls to or receive fax calls from endpoints on Avaya systems.

- Multiple hops and multiple conversions

  If a fax call must undergo more than one conversion cycle (from TDM protocol to IP protocol and back to TDM protocol), fax pass-through should be used. If fax relay mode is used, the call may fail due to delays in processing through more than one conversion cycle. A modem or TTY call may undergo no more than one conversion cycle (from TDM protocol to IP protocol and back to TDM protocol) on the communication path. If multiple conversion cycles occur, the call fails. As a result, both endpoint gateways and any intermediate servers in a path containing multiple hops must support shuffling for a modem, or TTY call to succeed.

  For example, in Figure 8: Shuffling for fax, modem, and TTY calls over IP on page 116, a hop occurs in either direction for calls between Port Network A and Media Gateway C because the calls are routed through Port Network D. In this case, shuffling is required on Port Network A for calls going to Media Gateway C, and shuffling is required on Port Network D for calls going from Media Gateway C to Port Network A.

Figure 8: Shuffling for fax, modem, and TTY calls over IP


Modem
Fax
TTY
A
Media Server with gateway/ port network
LAN/WAN
Inter-gateway IP connection
Media Gateway
B
Media Gateway
C
LAN/WAN
Inter-gateway IP connection
D
Media Server with gateway/ port network
Modem
Fax
TTY

Bandwidth for fax, modem, and TTY calls over IP networks

The following table identifies the bandwidth of fax, modem, and TTY calls based on packet sizes used, redundancy used, and whether the relay or pass through method is used.

Table 13: Bandwidth for fax, modem, and TTY calls over IP networks

Packet Size (in msec	Bandwidth (in kbps) (bidirectional)[1]									
	Redundancy = 0						Redundancy = 1		Red. = 2	Red. = 3
	TTY at G.711	TTY at G.729	TTY at G.723[2]	Fax Relay[3]	Modem Relay at 9600 Baud[4]	Fax/ Modem Pass Through[5]	Fax Relay[3, 4]	Fax/ Modem Pass Through	Fax Relay[3, 4]	Fax Relay[3, 4]
10	110	54	-	-	-	110	-	221	-	-
20	87	31	-	-	-	87	-	174	-	-
30	79	23	22	25	22.9	-	50	-	75	100
40	76	20	-	-	19.6	-	-	-	-	-
50	73	17	-	-	17.6	-	-	-	-	-
60	72	16	14	-	16.3	-	-	-	-	-

1. TTY, Modem Relay, Modem pass-through and Fax pass-through calls are full duplex. Multiply the mode's bandwidth by 2 to get the network bandwidth usage.
2. TTY at G723 supports packet size 30 and 60 ms.
3. Fax Relay supports packet size 30ms.
4. Non-zero redundancy options increase the bandwidth usage by a linear factor of the bandwidth usage when the redundancy is zero.
5. Fax and Modem Pass through supports packet sizes 10 and 20 ms.

AES/AEA Media Encryption

If media encryption is configured, the algorithm used during the audio channel setup of the call will be maintained for most fax relay and pass-through modes (seeEncryption options on page 1184). The exception is the T.38 Standard for Fax over IP. In this mode, encryption will not be used. Note that encrypted calls use 25% more Digital Signal Processing (DSP) capacity than non-encrypted calls.

Encryption is applicable as shown in the following table.

Table 14: Encryption options

Call Type	AEA	AES
Modem Pass-through	Y	Y
Modem Relay	Y	N
Fax Pass-through	Y	Y
Fax Relay	Y	N
TTY Pass-through	Y	Y
TTY Relay	Y	Y
T.38 Fax Standard	N	N

If the audio channel is encrypted, the fax digital channel is also encrypted except for the limitations described above. AEA-encrypted fax and modem relay calls that switch back to audio continue to be encrypted using the same key information used at audio call setup.

For the cases of encrypting fax, modem, and TTY pass through and TTY relay, the encryption used during audio channel setup is maintained for the call's duration.

The software behaves in the following way for encryption:

1. For fax, modem, and TTY pass through and relay, the VoIP firmware encrypts calls as administered on the codec set form. These calls begin in voice, so voip encrypts the voice channel as administered. If the media stream is converted to fax, modem, or TTY digital, the VoIP firmware automatically disables encryption as appropriate. When the call switches back to audio, VoIP firmware encrypts the stream again.
2. For T.38 fax, the VoIP firmware encrypts the voice channel as administered on the codec set form. When the call is converted to fax, the VoIP firmware automatically turns off encryption. If the call later reverts back to audio, VoIP firmware encrypts the stream again.

Chapter 3: Network quality administration

This section provides information for improving voice quality by adjusting the voice packet traffic behavior through an IP network, also known as implementing Quality of Service (QoS). The section covers these topics:

- About factors causing voice degradation introduces the types of voice degradation and their causes.
- About Quality of Service (QoS) and voice quality administration tells you how to administer your Avaya equipment for better voice quality and offers suggestions for other network problems.
- The About Media Encryption section discusses media encryption capabilities, requirements, and administration in Communication Manager.
- The About network management section includes information about administering H.248 Link Recovery and the Avaya Policy Manager (APM) and Avaya VoIP Monitoring Manager network monitoring tools.

Note:

Implementing QoS requires administration adjustments to Avaya equipment as well as LAN/WAN equipment (switches, routers, hubs, etc.).

For more information on implementing QoS, see the White Paper, *Avaya IP Voice Quality Network Requirements (EF-LB1500)*, at http://www1.avaya.com/enterprise/whitepapers/ip_networking.pdf.

About factors causing voice degradation

VoIP applications put severe constraints on the amount of end-to-end transfer delay of the voice signal and routing. If these constraints are not met, users complain of garbled or degraded voice quality, gaps, and pops. Due to human voice perception, VoIP applications can afford to randomly lose a few voice packets and the user can still understand the conversation. However, if voice packets are delayed or systematically lost, the destination experiences a momentary loss of sound, often with some unpleasing artifacts like clicks or pops. Some of the general complaints and their causes are listed in Table 15: User complaints and their causes on page 120.

Table 15: User complaints and their causes

Complaint	Possible causes and links to information
‘Talking over’ the far end	● Packet delay and loss ● Echo ● Network architecture between endpoint and intermediate node ● Switching algorithms
Near-end/ far-end hear(s) echo	● Impedance mismatch ● Improper coupling ● Codec administration
Voice is too soft or too loud	● PSTN loss ● Digital loss ● Automatic Gain Control ● Conference loss plan
Clicks, pops, or stutters	● Packet loss ● Timing drift due to clocks ● Jitter ● False DTMF detection ● Silence suppression algorithms
Voice sounds muffled, distorted, or noisy	● Codec administration ● Transducers ● Housings ● Environment ● Analog design

Some of the factors causing voice degradation are:

- Packet delay and loss
- Echo
- Transcoding
- Transcoding

Packet delay and loss

The causes of voice degradation include:

- Packet delay (latency)
    - Buffer delays
    - Queuing delays in switches and routers
    - Bandwidth restrictions
- Jitter (statistical average variance in end-to-end packet travel times)
- Packet loss
    - Network overloaded
    - Jitter buffers filled
    - Echo

For a detailed discussion of packet delay and loss, see the section on "Voice quality network requirements" in *Avaya Application Solutions: IP Telephony Deployment Guide*, 555-245-600.

Tip:

Avaya recommends a network assessment that measures and solves latency issues before implementing VoIP solutions (see *Avaya Application Solutions: IP Telephony Deployment Guide*, 555-245-600).

Echo

When you hear your own voice reflected back with a slight delay, this is echo and it happens for the following reasons:

- Electrical -- from unbalanced impedances or cross-talk
- Acoustical -- introduced by speakerphone or room size

The total round-trip time from when a voice packet enters the network to the time it is returned to the originator is echo path delay. In general, calls over a WAN normally have a longer echo path delay compared to calls over a LAN.

Note:

VoIP itself is not a cause of echo. However, significant amounts of delay and/or jitter associated with VoIP can make echo perceptible that would otherwise not be perceived.

Echo cancellers

Echo cancellers minimize echo by comparing the original voice pattern with the received patterns, and canceling the echo if the patterns match. However echo cancellers are not perfect, especially:

- When the round-trip delay from the echo canceller to the echo reflection point and back is longer than the time that the original (non-echoed) signal is buffered in the echo canceller memory. The larger the echo canceller's memory the longer the signal is held in the buffer, maximizing the number of packets that the canceller can compare in the allotted time.
- During Voice Activity Detection (VAD), which monitors the level of the received signal:
  - An energy drop of at least 3dB weaker than the original signal indicates echo.
  - An energy level 3dB greater indicates far-end speech.

Echo cancellers do not work well over analog trunks and with speakerphones with volume controls that permit strong signals. Although VADs can greatly conserve bandwidth, overly-aggressive VADs can cause voice clipping and reduce voice quality. VAD administration is done on the **station** form for the particular IP phone.

Analog trunks in IP configurations need careful network balance settings to minimize echo. A test tone of known power is sent out and the return signal measured to determine the balance setting, which is critical for reducing echo on IP calls across these trunks.

Echo cancellation plans

Communication Manager has several echo cancellation plans, as described in Table 16: Communication Manager echo cancellation plans on page 122.

Table 16: Communication Manager echo cancellation plans ***1 of 2***

Echo cancellation plan	Description	Use
EC-1	Inserts 6dB loss on the transmit side of the echo path. This results in asymmetrical transmit and receive sides. However, the 6dB loss is not needed on the receive side, since the signal is normally weaker than the transmit side's.	Used only when the other echo cancellation plans do not sufficiently reduce the echo to satisfactory levels.
EC-2		Use if speech is often clipped when both parties talk at the same time.
		1 of 2

Table 16: Communication Manager echo cancellation plans ***2 of 2***

Echo cancellation plan	Description	Use
EC-3		For slightly slower adaptation to echo, may result in a 2 or 3 second fade on strong echo for quiet talkers. Completely removes speech clipping.
EC-4	Same as EC-1 but without the 6dB loss in the echo path.	For very loud talkers and severe echo, the far-end talker's speech is heard as clipped when both parties talk at the same time.
		2 of 2

Transcoding

When IP endpoints are connected through more than one network region, it is important that each region use the same CODEC, the circuitry that converts an audio signal into its digital equivalent and assigns its companding properties. Packet delays occur when different CODECs are used within the same network region. In this case the IP Media Processor acts as a gateway translating the different CODECs, and an IP-direct (shuffled) connection is not possible.

Bandwidth

In converged networks that contain coexistent voice and data traffic, the volume of either type of traffic is unpredictable. For example, transferring a file using the File Transfer Protocol (FTP) can cause a sharp burst in the network traffic. At other times there may be no data in the network.

While most data applications are insensitive to small delays, the recovery of lost and corrupted voice packets poses a significant problem. For example, users might not really be concerned if the reception of E-mail or files from file transfer applications is delayed by a few seconds. In a voice call, the most important expectation is the real-time exchange of speech. To achieve this the network resources are required for the complete duration of the call. If in any instance, there are no resources or the network too busy to carry the voice packets, then the destination experiences clicks, pops and stutters. Therefore, there is a continuous need for a fixed amount of bandwidth during the call to keep it real-time and clear.

About Quality of Service (QoS) and voice quality administration

Of the VoIP network issues described in the About factors causing voice degradation section, delay is the most crucial. And because many of the other causes are highly interdependent with delay, the primary goal is to reduce delay by improving the routing in the network, or by reducing the processing time within the end points and the intermediate nodes.

For example, when delay is minimized:

- Jitter and electrically-induced echo abate.
- Intermediate node and jitter buffer resources are released making packet loss insignificant.

  As packets move faster in the network, the resources at each node are available for the next packet that arrives, and packets will not be dropped because of lack of resources.

Delay cannot be eliminated completely from VoIP applications, because delay includes the inevitable processing time at the endpoints plus the transmission time. However, the delay that is caused due to network congestion or queuing can be minimized by adjusting these Quality of Service (QoS) parameters:

- Layer 3 QoS
  - DiffServ
  - RSVP
- Layer 2 QoS: 802.1p/Q

These parameters are administered on the **IP Network Region** form (see Administering IP network regions on page 133).

Layer 3 QoS

DiffServ

The Differentiated Services Code Point (DSCP) or “DiffServ” is a packet prioritization scheme that uses the Type of Service (ToS) byte in the packet header to indicate the packet’s forwarding class and Per Hop Behaviors (PHBs). After the packets are marked with their forwarding class, the interior routers and gateways use this ToS byte to differentiate the treatment of packets.

A DiffServ policy must be established across the entire IP network, and the DiffServ values used by Communication Manager and by the IP network infrastructure must be the same.

If you have a Service Level Agreement (SLA) with a service provider, the amount of traffic of each class that you can inject into the network is limited by the SLA. The forwarding class is directly encoded as bits in the packet header. After the packets are marked with their forwarding class, the interior nodes (routers & gateways) can use this information to differentiate the treatment of packets.

RSVP

Resources ReSerVation Protocol (RSVP) can be used to lower DiffServ priorities of new calls when bandwidth is scarce. The RSVP signaling protocol transmits requests for resource reservations to routers on the path between the sender and the receiver for the voice bearer packets only, not the call setup or call signaling packets.

Layer 2 QoS: 802.1p/Q

802.1p is an Ethernet tagging mechanism that can instruct Ethernet switches to give priority to voice packets.

> ⚠ **CAUTION:**
>
> If you change 802.1p/Q on the IP Network Region screen, it changes the format of the Ethernet frames. 802.1p/Q settings in Communication Manager must match similar settings in your network elements.

The 802.1p feature is important to the endpoint side of the network since PC-based endpoints must prioritize audio traffic over routine data traffic.

IEEE standard 802.1Q allows you to specify both a virtual LAN (VLAN) and a frame priority at layer 2 for LAN switches or Ethernet switches, which allows for routing based on MAC addresses.

802.1p/Q provides for 8 priority levels and for a large number of Virtual LAN identifiers. Interpretation of the priority is controlled by the Ethernet switch and is usually based on highest priority first. The VLAN identifier permits segregation of traffic within Ethernet switches to reduce traffic on individual links. 802.1p operates on the MAC layer. The switch always sends the QoS parameter values to the IP endpoints. Attempts to change the settings by DHCP or manually are overwritten. The IP endpoints ignore the VLAN on/off options, because turning VLAN on requires that the capabilities be administered on the closet LAN switch nearest the IP endpoint. VLAN tagging can be turned on manually, by DHCP, or by TFTP.

If you have varied 802.1p from LAN segment to LAN segment, then you must administer 802.1p/Q options individually for each network interface. This requires a separate network region for each network interface.

Using VLANs

Virtual Local Area Networks (VLANs) provide security and create smaller broadcast domains by using software to create virtually-separated subnets. The broadcast traffic from a node that is in a VLAN goes to all the nodes that are members of this VLAN. This reduces CPU utilization and increases security by restricting the traffic to a few nodes rather than every node on the LAN.

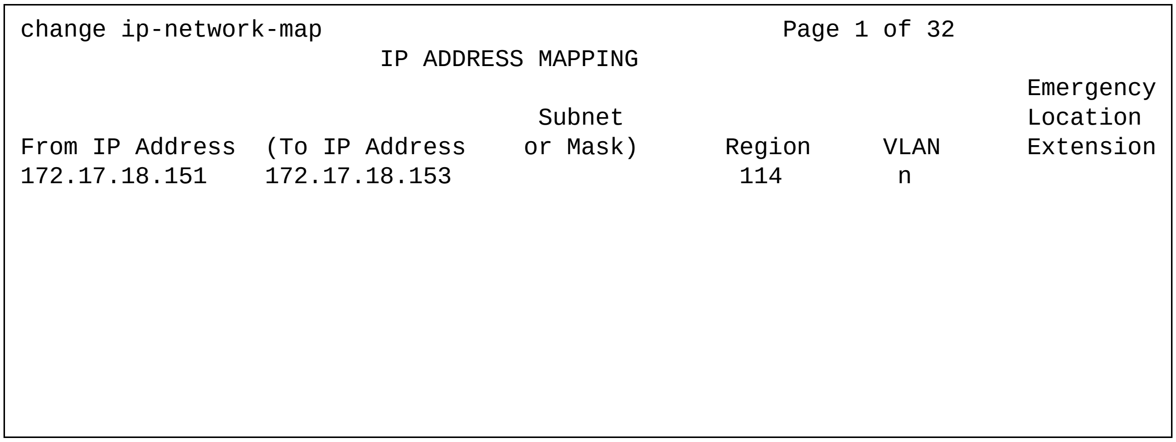
Any end-system that performs VLAN functions and protocols is “VLAN-aware,” although currently very few end-systems are VLAN-aware. VLAN-unaware switches cannot handle VLAN packets (from VLAN-aware switches), and this is why Avaya’s gateways have VLAN configuration turned off by default.

Avaya strongly recommends creating separate VLANs for VoIP applications. VLAN administration is at two levels:

- Circuit pack-level administration on the **IP-Interfaces** form (see Defining IP interfaces on page 67)
- Endpoint-level administration on the **IP Address Mapping** form

To administer endpoints for IP address mapping

1. Type **`change ip-network-map`** and press **Enter** to display the IP Address Mapping screen.



```
change ip-network-map                                          Page 1 of 32
                         IP ADDRESS MAPPING
                                                                          Emergency
                                    Subnet                                Location
From IP Address   (To IP Address    or Mask)      Region      VLAN        Extension
172.17.18.151     172.17.18.153                    114         n
```

2. Complete the following fields:

Table 17: IP Address Mapping screen fields *1 of 2*

Field	Conditions/Comments
From IP Address	Defines the starting IP address. A 32-bit address (four decimal numbers, each in the range **0-255**).
To IP Address	Defines the termination of the IP address. If this field and the **Subnet Mask** field are blank when submitted, the address in the **From IP Address** field is copied into this field. A 32-bit address (four decimal numbers, each in the range **0-255**).

1 of 2

Table 17: IP Address Mapping screen fields *2 of 2*

Field	Conditions/Comments
Subnet or Mask	Specifies the mask to be used to obtain the subnet work identifier from the IP address. If this field is non-blank on submission, then: ● Mask applied to **From IP Address** field, placing zeros in the non-masked rightmost bits. This becomes the stored "From" address. ● Mask applied to **To IP Address** field, placing 1's in the non-masked rightmost bits. This becomes the stored "To" address. If this field and the **To IP Address** field are blank when submitted, the address in the **From IP Address** field is copied into the **To IP Address** field. Valid entries: **0-32**, or blank.
Region	Identifies the network region for the IP address range. Valid entries: **1-250** (Enter the network region number for this interface.)
VLAN	Sends VLAN instructions to IP endpoints such as IP telephones/IP Softphones. This field does not send instructions to the PROCR, CLAN, or Media Processor boards. Valid entries: **0-4095** (specifies the virtual LAN value); **n** (disabled).
	2 of 2

3. Submit the screen.

Administering IP CODEC sets

The **IP Codec Set** screen allows you to specify the type of CODEC used for voice encoding and companding, and compression/decompression. The CODECs on the **IP Codec Set** screen are listed in the order of preferred use. A call across a trunk between two systems is set up to use the first common CODEC listed.

Note:

The CODEC order must be administered the same for each system of an H.323 trunk connection. The set of CODECs listed does not have to be the same, but the *order* of the listed CODECs must.

The **IP Codec Set** screen allows you to define the CODECs and packet sizes used by each IP network region. You can also enable or disable silence suppression for each CODEC in the set. The screen dynamically displays the packet size in milliseconds (ms) for each CODEC in the set, based on the number of frames you administer per packet.

Finally, you use this screen to assign the following characteristics to a codec set:

- Whether or not endpoints in the assigned network region can route fax, modem, or TTY calls over IP trunks
- Which mode the system uses to route the fax, modem, or TTY calls
- Whether or not redundant packets will be added to the transmission for higher reliability and quality

These characteristics must be assigned to the codec set, and the codec set must be assigned to a network region for endpoints in that region to be able to use the capabilities established on this screen.

CAUTION:

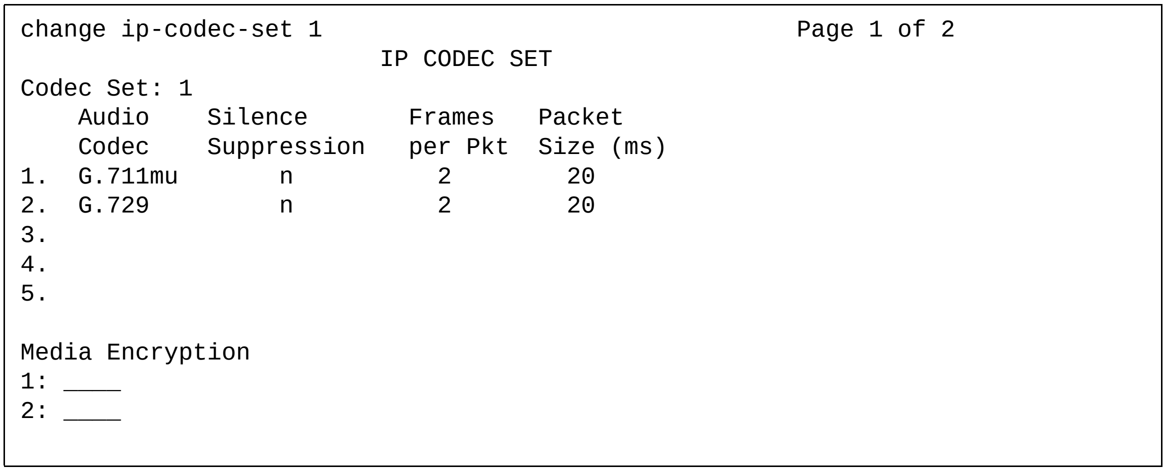
If users are using Super G3 fax machines as well as modems, do *not* assign these fax machines to a network region with an IP Codec set that is modem-enabled as well as fax-enabled. If its Codec set is enabled for both modem and fax signaling, a Super G3 fax machine incorrectly tries to use the modem transmission instead of the fax transmission.

Therefore, assign modem endpoints to a network region that uses a modem-enabled IP Codec set, and assign the Super G3 fax machines to a network region that uses a fax-enabled IP Codec set.

To administer an IP Codec set

1. Type **change ip-codec-set *set#*** and press **Enter** to open the **IP Codec Set** screen.

IP Codec Set screen, Page 1



```
change ip-codec-set 1                                        Page 1 of 2
                         IP CODEC SET
Codec Set: 1
    Audio    Silence       Frames   Packet
    Codec    Suppression   per Pkt  Size (ms)
1.  G.711mu       n          2        20
2.  G.729         n          2        20
3.
4.
5.

Media Encryption
1: ____
2: ____
```

2. Complete the fields inTable 18:

Note:

Use these approximate bandwidth requirements to decide which CODECs to administer. These numbers change with packet size, and do not include layer 2 overhead. With 20 ms packets the following bandwidth is required:

- G.711 A-law — 80 Kbps
- G.711 mu-law — 80 Kbps (used in U.S. and Japan)
- G.723.1-5.3 — 21.3 Kbps
- G.729 — 24 Kbps
- G.729B - 34.4 Kbps

Table 18: IP Codec Set screen fields, page 1 ***1 of 2***

Field	Conditions/Comments
Audio Codec	Specifies an audio CODEC. Valid values are: ● **G.711a** (a-law) ● **G.711mu (μ-law)** ● **G.723.1-5.3** ● **G.723.1-6.3** ● **G.729** ● **G.729B**
Silence Suppression	Enter **n** (recommended). Enter **y** if you require silence suppression on the audio stream. This may affect audio quality.
Frames per Pkt	Specifies frames per packet. Enter a value between **1-6**. Default values are: ● **2** for G.711 Codec (frame size 10ms) ● **2** for G729 Codec (frame size 10ms)

1 of 2

Table 18: IP Codec Set screen fields, page 1 ***2 of 2***

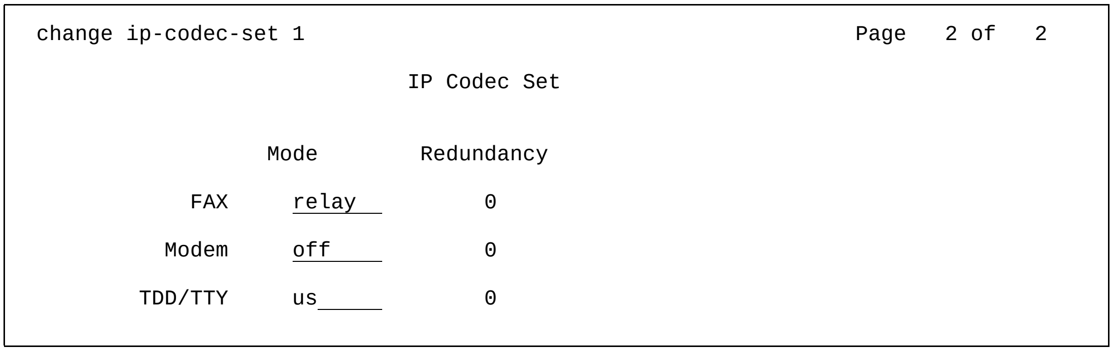
Field	Conditions/Comments
Packet Size (ms)	Automatically appears.
Media Encryption	This field appears only if the **Media Encryption over IP** feature is enabled. It specifies one of three possible options for the negotiation of encryption. The selected option for an IP codec set applies to all codecs defined in that set. Valid entries are: ● **aes** -- Advanced Encryption Standard (AES), a standard cryptographic algorithm for use by U.S. government organizations to protect sensitive (unclassified) information. Use this option to encrypt these links: - Server-to-gateway (H.248) - Gateway-to-endpoint (H.323) ● **aea** -- Avaya Encryption Algorithm. Use this option as an alternative to AES encryption when: - All endpoints within a network region using this codec set must be encrypted. - All endpoints communicating between two network regions and administered to use this codec set must be encrypted. ● **none** -- Media stream is unencrypted. This is the default setting.

2 of 2

3. Press **Next Page** to display page 2 of the screen.

   Page 2 appears.

IP-Codec-Set, page 2



```
change ip-codec-set 1                                          Page   2 of   2

                             IP Codec Set


                  Mode        Redundancy

            FAX     relay_____      0

          Modem     off_______      0

        TDD/TTY     us________      0
```

4. Complete the fields as described in the following table.

Table 19: IP Codec Set screen fields, page 2 *1 of 3*

Field	Conditions/Comments
FAX Mode	Specifies the mode for fax calls. Valid values are: ● **off** Turn off special fax handling when using this codec set. In this case, the fax is treated like an ordinary voice call. With a codec set that uses G.711, this setting is required to send faxes to non-Avaya systems that do not support T.38 fax. ● **relay** For users in regions using this codec, use Avaya relay mode for fax transmissions over IP network facilities. This is the default for new installations and upgrades to Communication Manager R2.1. ● **pass-through** For users in regions using this codec, use pass-through mode for fax transmissions over IP network facilities. This mode uses G.711-like encoding. ● **t.38-standard** For users in regions using this codec, use T.38 standard signaling for fax transmissions over IP network facilities.
	1 of 3

Table 19: IP Codec Set screen fields, page 2 ***2 of 3***

Field	Conditions/Comments
Modem Mode	Specifies the mode for modem calls. Valid values are: ● **off** Turn off special modem handling when using this codec set. In this case, the modem transmission is treated like an ordinary voice call. This is the default for new installations and upgrades to Communication Manager R2.1. With a codec set that uses G.711, this setting is required to send modem calls to non-Avaya systems. ● **relay** For users in regions using this codec, use relay mode for modem transmissions over IP network facilities. ● **pass-through** For users in regions using this codec, use pass-through mode for modem transmissions over IP network facilities.
TDD/TTY Mode	Specifies the mode for TDD/TTY calls. Valid values are: ● **off** Turn off special TTY handling when using this codec set. In this case, the TTY transmission is treated like an ordinary voice call. With a codec set that uses G.711, this setting is required to send TTY calls to non-Avaya systems. However, there may be errors in character transmissions. ● **US** For users in regions using this codec, use U.S. Baudot 45.45 mode for TTY transmissions over IP network facilities. This is the default for new installations and upgrades to Communication Manager R2.1. ● **UK** For users in regions using this codec, use U.K. Baudot 50 mode for TTY transmissions over IP network facilities. ● **pass-through** For users in regions using this codec, use pass-through mode for TTY transmissions over IP network facilities.

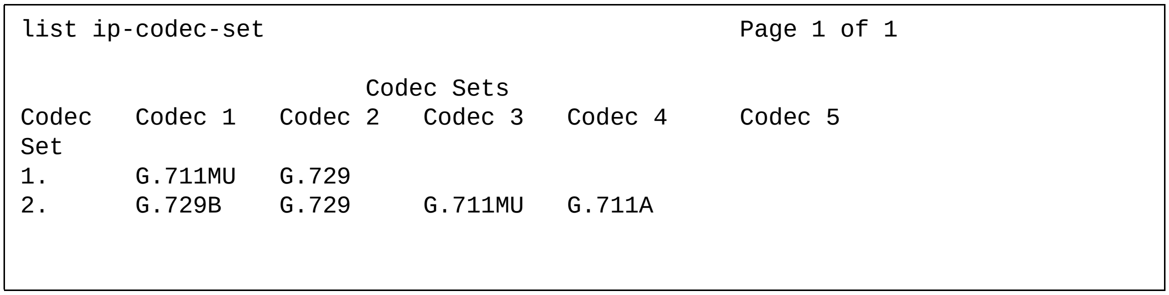
2 of 3

Table 19: IP Codec Set screen fields, page 2 ***3 of 3***

Field	Conditions/Comments
Redundancy	For each type of call (TTY, fax, or modem) that does *not* use pass-through mode, enter the number of duplicated packets, from **0** to **3**, that the system sends with each primary packet in the call. **0** means that you do not want to send duplicated packets. For any call types for which you selected pass-through mode, you can enter **0** or **1** only. That is, for pass-through mode, the maximum number of duplicated packets that the system can send with each primary packet is one.
	3 of 3

5. Submit the screen.
6. Type **`list ip-codec-set`** and press **Enter** to list all CODEC sets on the **CODEC Set** screen.

Codec Sets screen



```
list ip-codec-set                                   Page 1 of 1

                        Codec Sets
Codec   Codec 1   Codec 2   Codec 3   Codec 4     Codec 5
Set
1.      G.711MU   G.729
2.      G.729B    G.729     G.711MU   G.711A
```

7. Review your CODEC sets.

Administering IP network regions

Network regions enable you to group IP endpoints and/or VoIP and signaling resources that share the same characteristics. Signaling resources include Media Processor and C-LAN circuit packs. In this context, *IP endpoint* refers to IP stations, IP trunks, and G350 and G700 Media Gateways. The characteristics that can be defined for these IP endpoints and resources are:

- Audio Parameters
  - Codec Set
  - UDP port Range
  - Enabling Direct IP-IP connections
  - Enabling Hairpinning

- Quality of Service Parameters:
    - Diffserv settings
        - Call Control per-hop behavior (PHB)
        - VoIP Media PHB
    - 802.1p/Q settings
        - Call Control 802.1p priority
        - VoIP Media 802.1p priority
        - VLAN ID
    - Better than Best Effort (BBE) PHB
    - RTCP settings
    - RSVP settings
    - Location

The following sections tell you about:

- Defining an IP network region
- Interconnecting the network regions
- Reviewing the administration

Note:

For more information on using network regions, with examples, see the application note *Network Regions for Avaya MultiVantage™ Solutions - A Tutorial*, which is available at the Avaya Resource Library: http://www1.avaya.com/enterprise/resourcelibrary/applicationnotes/eclips_networking.html.

Defining an IP network region

⚠ **CAUTION:**

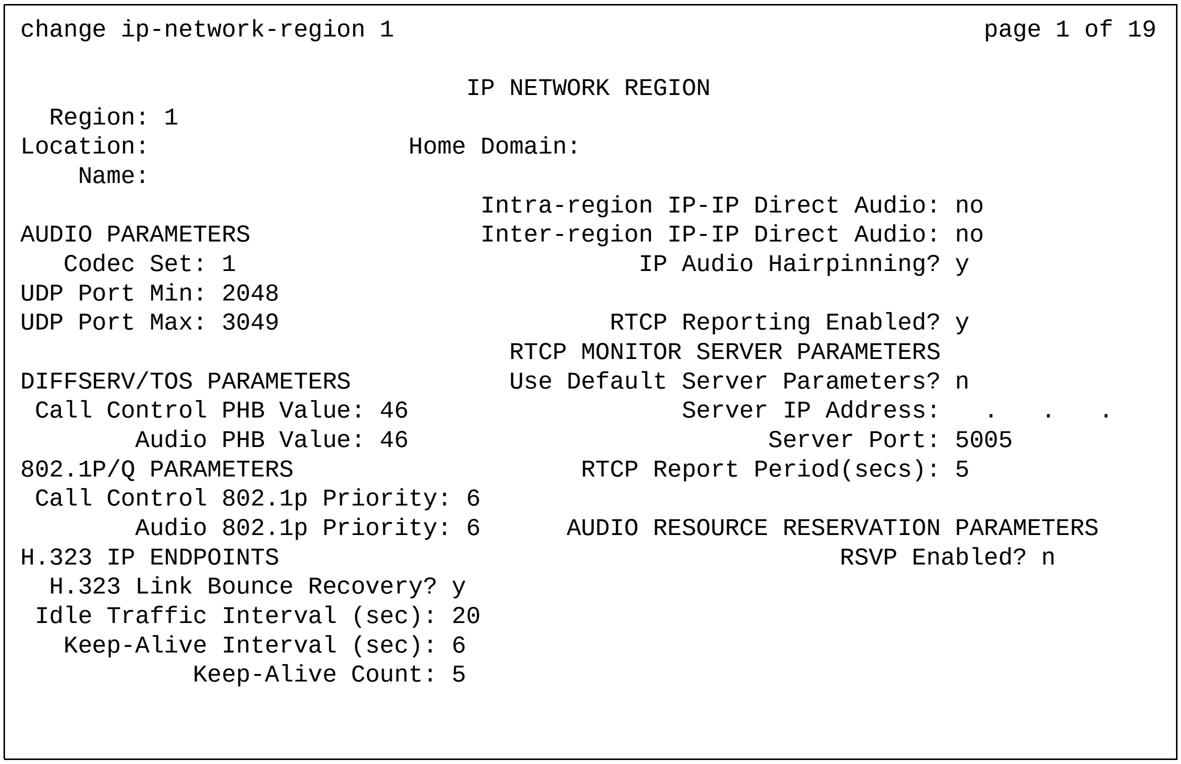
Never define a network region to span a WAN link.

Avaya strongly recommends that you accept the default values for the following screen.

To define an IP network region

1. Type **`change ip-network-region`** to open the **IP Network Region** screen.

IP Network Region screen



```
change ip-network-region 1                                         page 1 of 19

                               IP NETWORK REGION
  Region: 1
Location:                  Home Domain:
    Name:
                                Intra-region IP-IP Direct Audio: no
AUDIO PARAMETERS                Inter-region IP-IP Direct Audio: no
   Codec Set: 1                            IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 3049                       RTCP Reporting Enabled? y
                                  RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS           Use Default Server Parameters? n
 Call Control PHB Value: 46                   Server IP Address:   .   .   .
        Audio PHB Value: 46                         Server Port: 5005
802.1P/Q PARAMETERS                    RTCP Report Period(secs): 5
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 6
            Keep-Alive Count: 5
```

2. Complete the fields using the information in Table 20: IP Network Region field descriptions on page 135.

Table 20: IP Network Region field descriptions ***1 of 5***

Field	Descriptions/Comments
Region	Network Region number, **1–250**.
Location	Blank or **1–250**. Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and IP Softphones. **1-44** (DEFINITY R, CSI, SI only.) **1-250** S8300, S8500, S8700, S8710 **blank** The location is obtained from the cabinet containing the CLAN that the endpoint registered through, or the media gateway containing the Internal Call Controller or Local Survivable Processor on an Avaya S8300 Media Server through which the endpoint registered. This applies to IP telephones and IP Softphones. Traditional cabinets, Remote Offices, and the Avaya S8300 Media Server all have their locations administered on their corresponding screens.
	1 of 5

Table 20: IP Network Region field descriptions ***2 of 5***

Field	Descriptions/Comments
Name	Describes the region. Enter a character string up to 20 characters.
Home Domain	The network domain of the media server.
AUDIO PARAMETERS	
Codec Set	Specifies the CODEC set assigned to a region. Enter a value between **1-7** (default is **1**). **Note:** CODEC sets are administered on the **CODEC Set** screen (see Administering IP CODEC sets).
UDP Port-Min	Specifies the lowest port number to be used for audio packets. Enter a value between **2-65406** (default is **2048**). **Note:** This number must be twice the number of calls that you want to support plus one, must start with an even number, and must be consecutive. Minimum range is 128 ports. ⚠ **CAUTION:** Avoid the range of “well-known” or IETF-assigned ports. Do not use ports below 1024.
UDP Port-Max	Specifies the highest port number to be used for audio packets. Enter a value between **130-65535** (default is **65535**). ⚠ **CAUTION:** Avoid the range of well-known or IETF-assigned ports. Do not use ports below 1024.
DIFFSERVE/TOS PARAMETERS	
Call Control PHB Value	The decimal equivalent of the Call Control PHB value. Enter a value between **0-63**. ● Use PHB **46** for expedited forwarding of packets. ● Use PHB **46** for audio for legacy systems that only support IPv4 Type-of-Service, which correlates to the older ToS critical setting. ● Use PHB **46** if you have negotiated a Call Control PHB value in your SLA with your Service Provider.
	2 of 5

Table 20: IP Network Region field descriptions ***3 of 5***

Field	Descriptions/Comments
Audio PHB Value	The decimal equivalent of the VoIP Media PHB value. Enter a value between **0-63**: ● Use PHB **46** for expedited forwarding of packets. ● Use PHB **46** for audio for legacy systems that only support IPv4 Type-of-Service, which correlates to the older ToS critical setting.
802.1p/Q PARAMETERS	
Call Control 802.1p Priority	Specifies the 802.1p priority value, and appears only if the **802.1p/Q Enabled** field is **y**. The valid range is **0–7**. Avaya recommends **6** (high). See “Caution” below this table.
Audio 802.1p Priority	Specifies the 802.1p priority value, and appears only if the **802.1p/Q Enabled** field is **y**. The valid range is **0–7**. Avaya recommends **6** (high). See “Caution” below this table.
H.323 IP ENDPOINTS	
H.323 Link Bounce Recovery	**y/n** Specifies whether to enable H.323 Link Bounce Recovery feature for this network region.
Idle Traffic Interval (sec)	**5-7200** Enter the maximum traffic idle time in seconds. Default is **20**.
Keep-Alive Interval (sec)	**1-120** Specify the interval between KA retransmissions in seconds. Default is **5**.
Keep-Alive Count	**1-20** Specify the number of retries if no ACK is received. Default is **5**.
Intra-region IP-IP Direct Audio	**y/n** Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions. Enter **native (NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is that of the IP telephone/IP Softphone itself (without being translated by NAT). IP phones must be configured behind a NAT device *before* this entry is enabled. Enter **translated (NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections within the region is to be the one with which a NAT device replaces the native address. IP phones must be configured behind a NAT device *before* this entry is enabled.
	3 of 5

Table 20: IP Network Region field descriptions ***4 of 5***

Field	Descriptions/Comments
Inter-region IP-IP Direct Audio	**y/n** Enter **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions. Enter **translated (NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is to be the one with which a NAT device replaces the native address. IP phones must be configured behind a NAT device *before* this entry is enabled. Enter **native (NAT)** if the IP address from which audio is to be received for direct IP-to-IP connections between regions is that of the telephone itself (without being translated by NAT). IP phones must be configured behind a NAT device *before* this entry is enabled.
IP Audio Hairpinning?	**y/n** Enter **y** to allow IP endpoints to be connected through the media server's IP circuit pack in IP format, without first going through the Avaya TDM bus.
RTCP Reporting Enabled?	Specifies whether you want to enable RTCP reporting. If this field is set to **y**, then the RTCP Monitor Server Parameters fields appear.
RTCP MONITOR SERVER PARAMETERs	
Use Default Server Parameters?	This field only appears when the **RTCP Reporting Enabled** field is set to **y**. ● Enter **y** to use the default RTCP Monitor server parameters as defined on the IP Options System Parameters screen. If set to **y**, you must complete the **Default Server IP Address** field on the **IP Options System Parameters** screen (**`change system-parameters ip-options`**). ● If you enter **n**, you need to complete the **Server IP Address**, **Server Port**, and **RTCP Report Period** fields.
Server IP Address	This field only appears when the **Use Default Server Address** field is set to **n** and the **RTCP Enabled** field is set to **y**. Enter the IP address for the RTCP Monitor server in **nnn.nnn.nnn.nnn** format, where **nnn=0-255.**
Server Port	This field only appears when the **Use Default Server Address** field is set to **n** and the **RTCP Enabled** field is set to **y**. Enter the port (**1-65535**) for the RTCP Monitor server.
RTCP Report Period (sec)	This field only appears when the **Use Default Server Address** field is set to **n** and the and the **RTCP Enabled** field is set to **y**. Range of values is **5-30** (seconds).
	4 of 5

Table 20: IP Network Region field descriptions ***5 of 5***

Field	Descriptions/Comments
AUDIO RESOURCE RESERVATION PARAMETERS	
RSVP Enabled?	**y/n** Specifies whether or not you want to enable RSVP.
RSVP Refresh Rate (sec)	Enter the RSVP refresh rate in seconds (**1-99**). This field only appears if the **RSVP Enabled** field is set to **y**.
Retry upon RSVP Failure Enabled	Specifies whether to enable retries when RSVP fails (**y/n**). This field only appears if the **RSVP Enabled** field is set to **y**.
RSVP Profile	This field only appears if the **RSVP Enabled** field is set to **y**. You set this field to what you have configured on your network ● **guaranteed-service** places a limit on the end-to-end queuing delay from the sender tot he receiver. This is the most appropriate setting for VoIP applications. ● **controlled-load** (a subset of **guaranteed-service**) provides for a traffic specifier but not the end-to-end queuing delay.
RSVP unreserved (BBE) PHB Value	Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop. Enter the decimal equivalent of the DiffServ Audio PHB value, **0-63**. This field only appears if the **RSVP Enabled** field is set to **y.** **Note:** The "per-flow state and signaling" is RSVP, and when RSVP is not successful, the BBE value is used to discriminate between Best Effort and voice traffic that has attempted to get an RSVP reservation, but failed.
	5 of 5

⚠ **CAUTION:**

If you change 802.1p/Q on the **IP Network Region** screen, it changes the format of the Ethernet frames. 802.1p/Q settings in Communication Manager must match those in all of the interfacing elements in your data network.

3. Press **Enter** to save the changes.

Interconnecting the network regions

If TN799DP (C-LAN) and TN2302AP (IP Media Processor) resources are shared between/among administered network regions, you must define which regions communicate with which other regions and with what CODEC set on the **Inter-Network Region Connection Management** form (`change/display/status ip-network-region`).

Note:

You cannot connect IP endpoints in different network regions or communicate between/among network regions unless you specify the CODEC set on this form.

You can also specify for the *Call Admission Control - Bandwidth Limitation* feature:

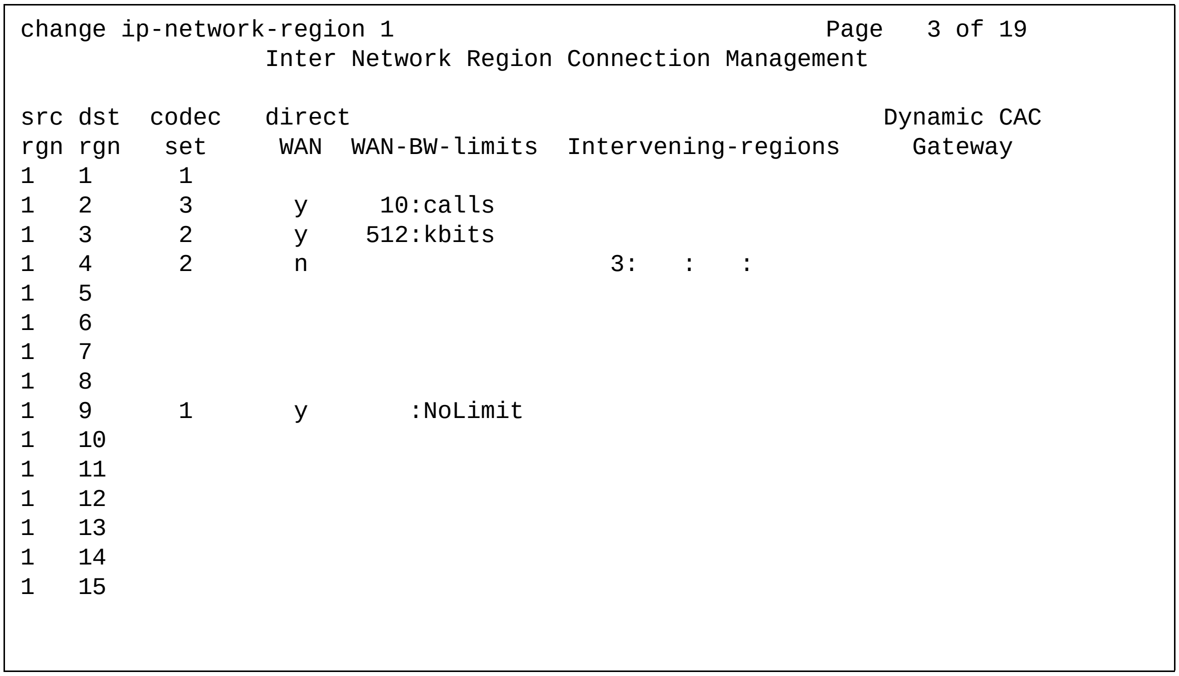
- Whether regions are directly connected or indirectly connected through intermediate regions.
- Bandwidth limits for IP bearer traffic between two regions using either a maximum bit rate or number of calls.

  When a bandwidth limit is reach, additional IP calls between those regions are diverted to other channels or blocked.

  Typically, the bandwidth limit is specified as the number of calls when the codec set administered across a WAN link contains a single codec. When the codec set administered across a WAN link contains multiple codecs, the bandwidth limit is usually specified as a bit-rate. For regions connected across a LAN, the normal bandwidth limit setting is **nolimit**.

To administer inter-network region connections

1. Type **`change ip-network-region`** ***`region#`*** and press **Enter** to open the **Inter Network Region Connection Management** screen.



```
change ip-network-region 1                              Page   3 of 19
                 Inter Network Region Connection Management

src dst  codec   direct                                      Dynamic CAC
rgn rgn   set     WAN  WAN-BW-limits  Intervening-regions      Gateway
1   1      1
1   2      3       y     10:calls
1   3      2       y    512:kbits
1   4      2       n                     3:   :   :
1   5
1   6
1   7
1   8
1   9      1       y       :NoLimit
1   10
1   11
1   12
1   13
1   14
1   15
```

2. Specify CODEC sets for your shared network regions by placing a CODEC set number in the **codec-set** column. Specify the type of inter-region connections and bandwidth limits in the remaining columns.

   In the example, network region 1 is directly connected to regions 2, 3, and 9, and is indirectly connected (through region 3) to region 4. Network region 1 communicates with the following other network regions using the specified CODEC sets and bandwidth limits:

   - Network region 1 using CODEC set 1
   - Network region 2 using CODEC set 3 with a 10-call bandwidth limit.
   - Network region 3 using CODEC set 2 with a 512 kbps bandwidth limit.
   - Network region 4 using CODEC set 2
   - Network region 9 using CODEC set 1 with no bandwidth limit.

3. Press **Enter** to save the changes.

Status of inter-region usage

You can check the status of bandwidth usage between network regions using: **`status ip-network-region`** ***`n`*** or ***`n/m`***. Using the ***`n`***, the connection status, bandwidth limits, and bandwidth usage is displayed for all regions directly connected to ***`n`***. For regions indirectly connected to ***`n`***, just the connection status is displayed. If regions ***`n`*** and ***`m`*** are indirectly connected, using ***`n/m`*** in the command displays the connection status, bandwidth limits, and bandwidth usage, for each intermediate connection.

To administer the network region on the Signaling Group form

Note:

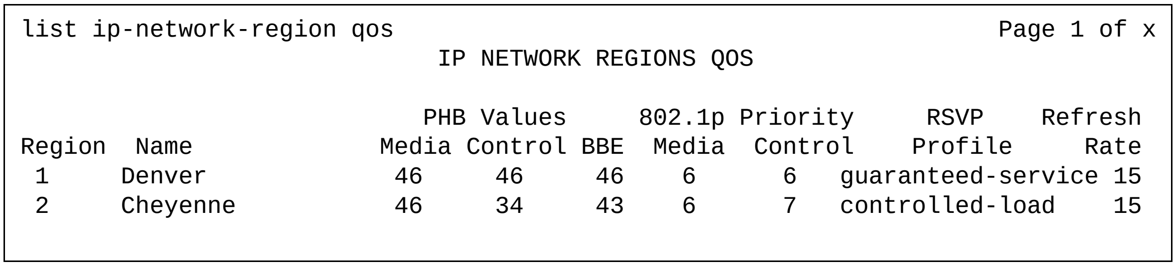
The S8300 media server does not support signaling groups.

1. Type **`change signaling-group`** ***`group#`*** and press **Enter** to display the **Signaling Group** form.
2. Type the number of the network region that corresponds to this signaling group in the **Far-end Network Region** field:
   - **1-80** (si)
   - **1-250** (r and S8300 or S8700 servers)
3. Press **Enter** to save the changes.

Reviewing the administration

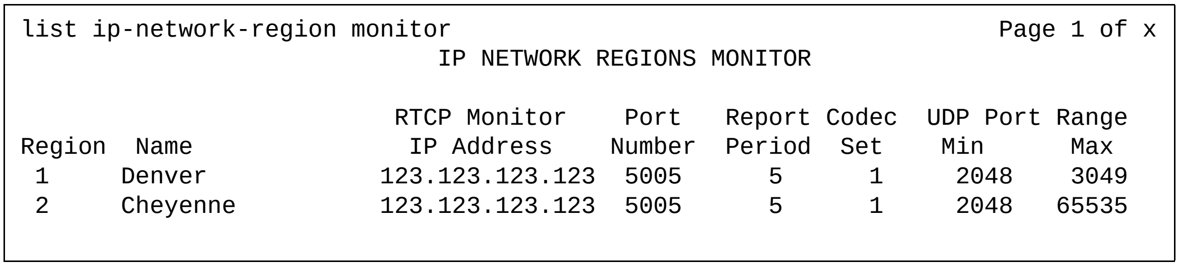
To check the network region administration:

1. Type **`list ip-network-region qos`** and press **Enter** to display the **IP Network Regions QOS** screen.



```
list ip-network-region qos                                    Page 1 of x
                             IP NETWORK REGIONS QOS

                            PHB Values     802.1p Priority     RSVP    Refresh
Region  Name             Media Control BBE  Media  Control    Profile     Rate
 1     Denver             46      46     46    6       6   guaranteed-service 15
 2     Cheyenne           46      34     43    6       7   controlled-load    15
```

2. Ensure that you have the proper values for each network region and that the regions are interconnected according to your design.
3. Type **`list ip-network-region monitor`** and press **Enter** to see the **IP Network Regions Monitor** screen, which includes information about the CODEC sets.



```
list ip-network-region monitor                                Page 1 of x
                             IP NETWORK REGIONS MONITOR

                          RTCP Monitor    Port   Report Codec  UDP Port Range
Region  Name               IP Address    Number  Period  Set    Min       Max
 1     Denver            123.123.123.123  5005      5      1     2048     3049
 2     Cheyenne          123.123.123.123  5005      5      1     2048    65535
```

4. Ensure that the audio transport parameters are administered according to your design.

Setting network performance thresholds

Note:

The *craft* (or higher) login is required to perform this administration.

Communication Manager gives you control over four IP media packet performance thresholds to help streamline VoIP traffic. You can use the default values for these parameters, or you can change them to fit the needs of your network. These threshold values apply only to IP trunks and do not affect other IP endpoints.

Note:

You cannot administer these parameters unless these conditions are met:

- The **Group Type** field on the **Signaling Group** form is **h.323** or **sip**.

- The **Bypass If IP Threshold Exceeded** field is set to **y** on the **Signaling Group** form.

  If bypass is activated for a signaling group, ongoing measurements of network activity collected by the system are compared with the values in the **IP-options system-parameters** screen. If the values of these parameters are exceeded by the current measurements, the bypass function terminates further use of the network path associated with the signaling group. The following actions are taken when thresholds are exceeded:

  - Existing calls on the IP trunk associated with the signaling group are not maintained.
  - Incoming calls are not allowed to arrive at the IP trunks on the bypassed signaling group and are diverted to alternate routes.
  - Outgoing calls are blocked on this signaling group.

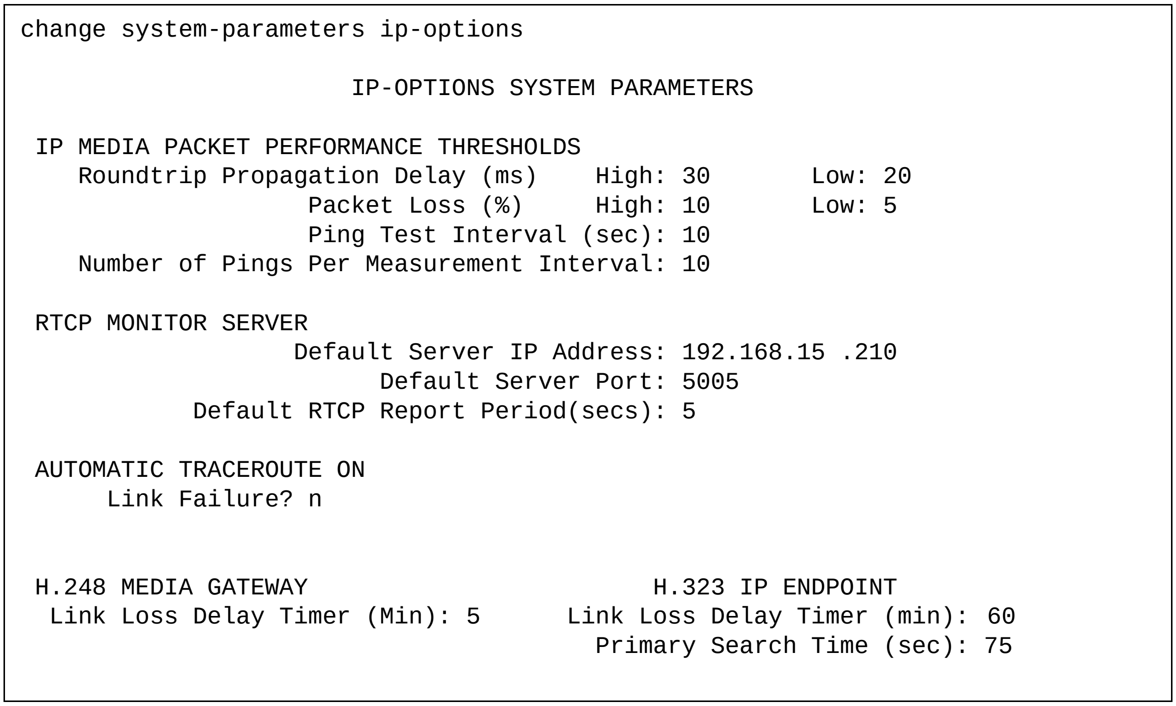
    If so administered, blocked calls are diverted to alternate routes (either IP or circuits) as determined by the administered routing patterns.

Note:

Avaya strongly recommends that you use the default values.

To administer network performance parameters

1. Enter **`change system-parameters ip-options`** to open the **IP Options System Parameters** screen.



```
change system-parameters ip-options

                       IP-OPTIONS SYSTEM PARAMETERS

 IP MEDIA PACKET PERFORMANCE THRESHOLDS
    Roundtrip Propagation Delay (ms)    High: 30       Low: 20
                    Packet Loss (%)     High: 10       Low: 5
                    Ping Test Interval (sec): 10
    Number of Pings Per Measurement Interval: 10

 RTCP MONITOR SERVER
                   Default Server IP Address: 192.168.15 .210
                         Default Server Port: 5005
            Default RTCP Report Period(secs): 5

 AUTOMATIC TRACEROUTE ON
      Link Failure? n


 H.248 MEDIA GATEWAY                        H.323 IP ENDPOINT
  Link Loss Delay Timer (Min): 5      Link Loss Delay Timer (min): 60
                                        Primary Search Time (sec): 75
```

2. Enter values for the fields suitable for your network needs (defaults shown in the table below).

Field	Conditions/
Roundtrip Propagation Delay (ms)	High: **800** Low: **400**
Packet Loss (%)	High: **40** Low: **15**
Ping Test Interval (sec)	**20**
Number of Pings per Measurement Interval	**10**

3. Press **Enter** to save the changes.

Enabling spanning tree (STP)

Spanning Tree (STP) is a loop avoidance protocol. If you don't have loops in your network, you don't need STP. The "safe" option is always to leave STP enabled. Failure to do so on a network with a loop (or a network where someone inadvertently plugs the wrong cable into the wrong ports) will lead to a complete cessation of all traffic.

STP is slow, however. It is slow to converge after a network failure, and slow to allow a new port into the network (~50 sec by default). In early generations of phones, this delay caused phone DHCP requests to time out, and the phones didn't recover gracefully, so there was a recommendation to use "port fast" (Cisco) or "fast start" (some Cajuns). Because Cajun P330s didn't support those features, the recommendation was modified to *disable* STP on phone ports on those switches. This soon came to be understood as disable STP everywhere, which was not necessary.

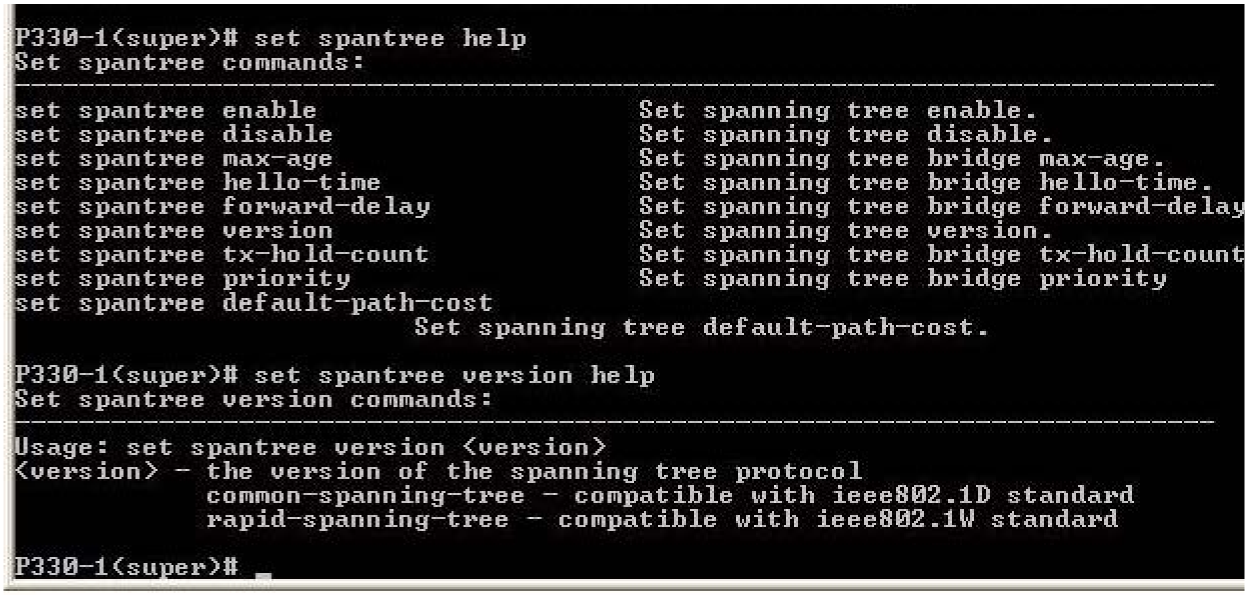
Since that time, Rapid Spanning Tree emerged, which converges faster than the earlier STP, and enables new ports much faster (sub-second) than the older protocol. **Rapid Spanning Tree** works with all Avaya equipment, and can be *recommended*. Spanning Tree is set using the P330 stack processor command line interface.

To enable/disable spanning tree

1. Open a telnet session on the P330 stack processor, using the serial cable connected to the Console port of the G700.
2. At the **P330-x(super)#** prompt, type **`set spantree help`** and press **Enter** to display the set spantree commands selection.

   The full set of Spanning Tree commands is displayed in Figure 9.

Figure 9: Set Spantree commands



```
P330-1(super)# set spantree help
Set spantree commands:
----------------------------------------------------------------------------
set spantree enable                    Set spanning tree enable.
set spantree disable                   Set spanning tree disable.
set spantree max-age                   Set spanning tree bridge max-age.
set spantree hello-time                Set spanning tree bridge hello-time.
set spantree forward-delay             Set spanning tree bridge forward-delay
set spantree version                   Set spanning tree version.
set spantree tx-hold-count             Set spanning tree bridge tx-hold-count
set spantree priority                  Set spanning tree bridge priority
set spantree default-path-cost
                         Set spanning tree default-path-cost.

P330-1(super)# set spantree version help
Set spantree version commands:
----------------------------------------------------------------------------
Usage: set spantree version <version>
<version> - the version of the spanning tree protocol
            common-spanning-tree - compatible with ieee802.1D standard
            rapid-spanning-tree - compatible with ieee802.1W standard

P330-1(super)# _
```

3. To enable Spanning Tree, type **`set spantree enable`** and press **Enter**.
4. To set the version of Spanning Tree, type **`set spantree version help`** and press **Enter**.

   The selection of Spanning Tree protocol commands displays (see Figure 9).
5. To set the **rapid spanning tree** version, type **`set spantree version rapid-spanning-tree`** and press **Enter**.

   The 802.1w standard defines differently the default path cost for a port compared to STP (802.1d). In order to avoid network topology change when migrating to RSTP, the STP path cost is preserved when changing the spanning tree version to RSTP. You can use the default RSTP port cost by typing the CLI command **`set port spantree cost auto`**.

   Note:

   > Avaya P330s now support a "Faststart" or "Portfast" function, because the 802.1w standard defined it. An edge port is a port that goes to a device that cannot form a network loop.
   > To set an **edge-port**, type **`set port edge admin state`** ***`module/port`*** **`edgeport`**.

For more information on the Spanning Tree CLI commands, see the *Avaya P330 User's Guide* (available at http://www.avaya.com/support).

Adjusting jitter buffers

Since network packet delay is usually a factor, jitter buffers should be no more than twice the size of the largest statistical variance between packets. The best solution is to have dynamic jitter buffers that change size in response to network conditions. Avaya equipment uses dynamic jitter buffers.

- Check for network congestion
- Bandwidth too small
- Route changes (can interact with network congestion or lack of bandwidth)

Configuring UDP ports

Communication Manager allows users to configure User Datagram Protocol (UDP) port ranges that are used by VoIP packets. Network data equipment uses these port ranges to assign priority throughout the network. Communication Manager can download default values to the endpoint when those values are not provided by the endpoint installer or the user.

About Media Encryption

This section provides information on the use and administration of Avaya Communication Manager Media Encryption. Use any of the following links to go to the appropriate section:

- What is Media Encryption?
- What limitations does Media Encryption have?
- What are the requirements for Media Encryption?
- Is there a license file requirement?
- Administering Media Encryption
- How does Media Encryption interact with other features?
- About legal wiretapping
- About possible failure conditions

What is Media Encryption?

To provide privacy for media streams that are carried over IP networks, Avaya Communication Manager supports encryption for IP bearer channel (RTP Audio) between any combination of media gateways and IP endpoints.

Digitally encrypting the audio (voice) portion of a VoIP call can reduce the risk of electronic eavesdropping. IP packet monitors, sometimes called sniffers, are to VoIP calls what wiretaps are to circuit-switched (TDM) calls, except that an IP packet monitor can watch for and capture unencrypted IP packets and can play back the conversation in real-time or store it for later playback.

With media encryption enabled, Communication Manager encrypts IP packets before they traverse the IP network. An encrypted conversation sounds like white noise or static when played through an IP monitor. End users do not know that a call is encrypted because there are:

- No visual or audible indicators to indicate that the call is encrypted.
- No appreciable voice quality differences between encrypted calls and non-encrypted calls.

What limitations does Media Encryption have?

⚠ **SECURITY ALERT:**
Be sure that you understand these important media encryption limitations:

1. Any call that involves a circuit-switched (TDM) endpoint such as a DCP or analog phone is vulnerable to conventional wire-tapping techniques.

2. Any call that involves an IP endpoint or gateway that does not support encryption can be a potential target for IP monitoring. Common examples are IP trunks to 3rd-party vendor switches.

3. Any party that is not encrypting an IP conference call exposes all parties on the IP call between the unencrypted party and its supporting media processor to monitoring, even though the other IP links are encrypting.

What are the requirements for Media Encryption?

Table 21: Media Encryption requirements on page 148 lists the supported hardware, software, and firmware requirements for Media Encryption.

Table 21: Media Encryption requirements

Hardware	Minimum Software or Firmware	
	AEA	AES
Communication Manager	CM1.3	CM 2.0
Avaya IP phones:		
4601	R1.8	N/A
4602	R1.8	N/A
4606	R1.8	N/A
4610SW	N/A	R2.0
4612	R1.8	N/A
4620	R1.8	R2.0
4620SW	N/A	R2.0
4624	R1.8	N/A
4630	R1.8	N/A
4690	N/A	N/A
IP Softphone	R4V1 with service pack 1	R5
IP SoftConsole	R1.5	R2
TN2302AP IP Media Processor circuit pack	V47	V47
IP Agent	R5	R5

Media Encryption does not work with the following devices:

- Any gateway or IP endpoint that cannot support the Avaya Encryption Algorithm (AEA)
- Any wired, circuit-switched (TDM) telephone (digital or analog) or trunk

Is there a license file requirement?

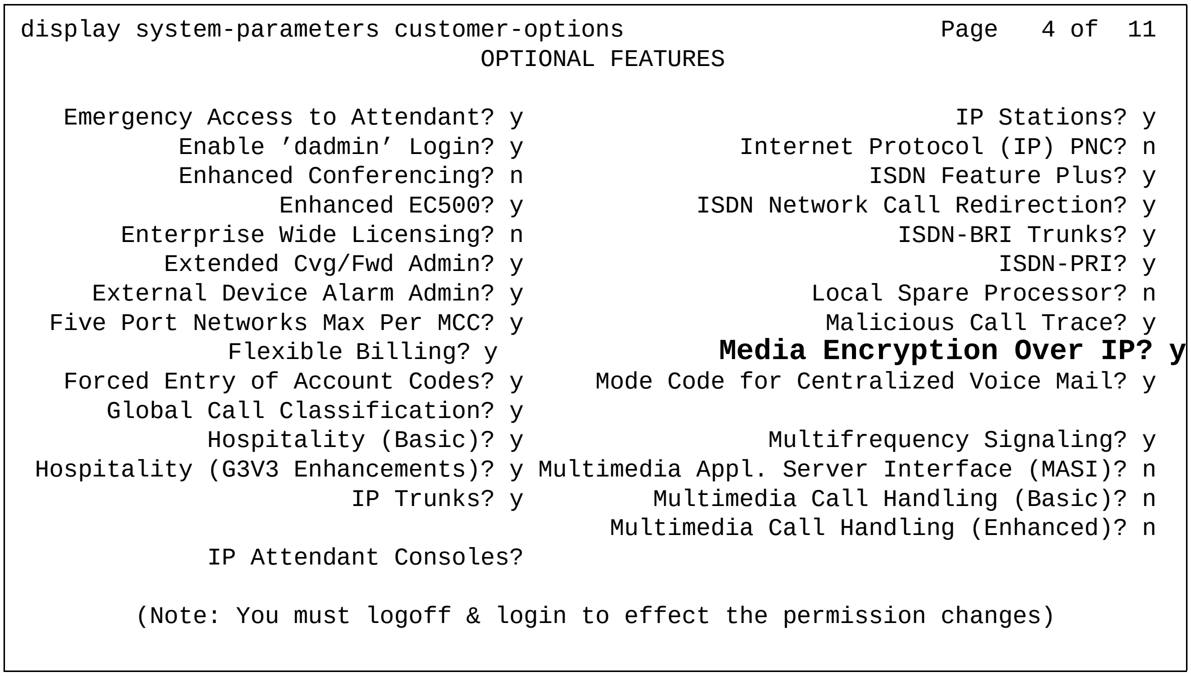
Media Encryption does not work unless the server has a valid license file with Media Encryption enabled. First check the current license file (Is Media Encryption currently enabled?) and if Media Encryption is not enabled, then you must install a license file with Media Encryption enabled.

Is Media Encryption currently enabled?

To determine whether Media Encryption is enabled in the current License File:

1. At the SAT type **`display system-parameters customer-options`** and press **Enter** to display the **Optional Features** form.
2. Scroll to page 4 and verify that the **Media Encryption Over IP?** field is *y*.

Media encryption field on Optional Features form



```
display system-parameters customer-options                      Page    4 of  11
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                              IP Stations? y
           Enable ’dadmin’ Login? y               Internet Protocol (IP) PNC? n
           Enhanced Conferencing? n                        ISDN Feature Plus? y
                  Enhanced EC500? y           ISDN Network Call Redirection? y
       Enterprise Wide Licensing? n                          ISDN-BRI Trunks? y
          Extended Cvg/Fwd Admin? y                                 ISDN-PRI? y
     External Device Alarm Admin? y                     Local Spare Processor? n
  Five Port Networks Max Per MCC? y                      Malicious Call Trace? y
              Flexible Billing? y                Media Encryption Over IP? y
   Forced Entry of Account Codes? y     Mode Code for Centralized Voice Mail? y
      Global Call Classification? y
             Hospitality (Basic)? y                  Multifrequency Signaling? y
 Hospitality (G3V3 Enhancements)? y Multimedia Appl. Server Interface (MASI)? n
                      IP Trunks? y          Multimedia Call Handling (Basic)? n
                                         Multimedia Call Handling (Enhanced)? n
             IP Attendant Consoles?

        (Note: You must logoff & login to effect the permission changes)
```

Media Encryption is enabled by default in the U. S. and other countries unless prohibited by export regulations.

Administering Media Encryption

This section contains Avaya Communication Manager administration procedures for:

- Administering Media Encryption for IP codec sets
- Administering Media Encryption for signaling groups

Note:

IP endpoints do not require any encryption administration, and end users do not have to do anything to use media encryption.

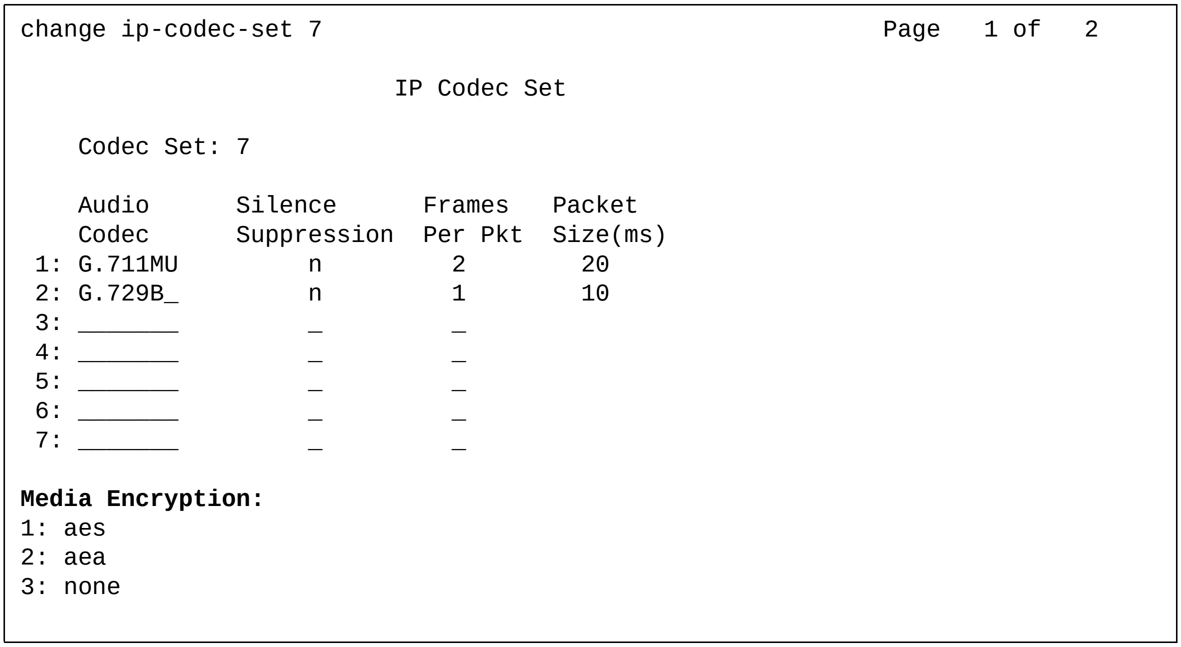
Administering Media Encryption for IP codec sets

The **IP Codec Set** form allows you to independently administer codec sets to use media encryption or not.

To administer media encryption on all codecs in an IP codec set:

1. At the SAT type **change ip-codec-set *number*** and press **Enter** to display the **IP Codec Set** form.

Media Encryption field on the IP Codec Set form



```
change ip-codec-set 7                                       Page   1 of   2

                          IP Codec Set

    Codec Set: 7

    Audio      Silence      Frames   Packet
    Codec      Suppression  Per Pkt  Size(ms)
 1: G.711MU         n         2        20
 2: G.729B_         n         1        10
 3: _______         _         _
 4: _______         _         _
 5: _______         _         _
 6: _______         _         _
 7: _______         _         _

Media Encryption:
1: aes
2: aea
3: none
```

2. Administer the **Media Encryption** field to one of the values in Table 22: Media Encryption Field Values (IP Codec Set) on page 151:

   Note:

   > The option that you select for the **Media Encryption** field for each codec set applies to all codecs defined in that set.

   Note:

   > This field is hidden if the **Media Encryption Over IP?** field on the **Customer Options** form (Media encryption field on Optional Features form on page 149) is *n*. The **Media Encryption** field appears only if the **Media Encryption over IP** feature is enabled in the license file (and displays as *y* on the **Customer Options** form).

The **Media Encryption** field specifies one, two, or three options for the negotiation of encryption — **aes**, **aea**, and **none**. The order in which the options are listed signifies the preference of use, similar to the list of codecs in a codec set. Two endpoints must support at least one common encryption option for a call to be completed between them.

The selected options for an IP codec set applies to all codecs defined in that set.

Table 22: Media Encryption Field Values (IP Codec Set)

Valid entries	Usage
aes	Advanced Encryption Standard (AES), a standard cryptographic algorithm for use by U.S. government organizations to protect sensitive (unclassified) information. AES reduces circuit-switched-to-IP call capacity by 25%.
aea	Avaya Encryption Algorithm. AEA is not as secure an algorithm as AES but call capacity reduction with AEA is negligible. Use this option as an alternative to AES encryption when: ● All endpoints within a network region using this codec set must be encrypted. ● All endpoints communicating between two network regions and administered to use this codec set must be encrypted.
none	Media stream is unencrypted. This option prevents encryption when using this codec set and is the default setting when Media Encryption is not enabled.

Note:

> The initial default value for this field is *none* when the **Media Encryption Over IP?** field in the **Optional Features** screen (on the **Customer Options** form) is enabled (*y*) for the first time. If this field is *n*, the **Media Encryption** field on the **IP Codec Set** screen is hidden and functions as if *none* was selected.

The following table lists the mapping between the Media Encryption values used in Communication Manager 1.3 and 2.x.

Table 23: Media Encryption field values (IP Codec Set form)

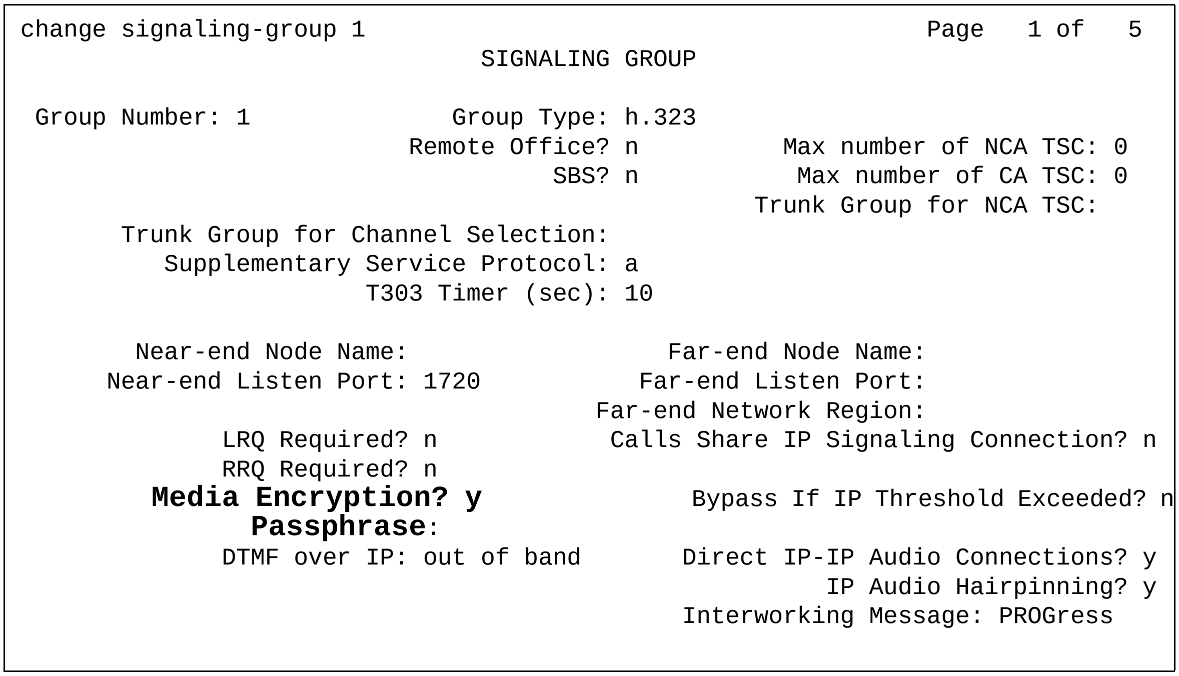
Communication Manager 1.3 Field Value	Communication Manager 2.x Field Value
always	aea
preferred	1. aea 2. none
optional	1. none 2. aea
never	none

Administering Media Encryption for signaling groups

To administer Media Encryption for an IP signaling group:

1. At the SAT type **`change signaling-group`** ***`number`*** to display the **Signaling Group** form

Media encryption and passphrase fields for signaling groups



```
change signaling-group 1                                       Page   1 of   5
                                SIGNALING GROUP

 Group Number: 1             Group Type: h.323
                          Remote Office? n         Max number of NCA TSC: 0
                                    SBS? n          Max number of CA TSC: 0
                                                  Trunk Group for NCA TSC:
       Trunk Group for Channel Selection:
          Supplementary Service Protocol: a
                        T303 Timer (sec): 10

        Near-end Node Name:                  Far-end Node Name:
      Near-end Listen Port: 1720           Far-end Listen Port:
                                        Far-end Network Region:
              LRQ Required? n            Calls Share IP Signaling Connection? n
              RRQ Required? n
         Media Encryption? y                 Bypass If IP Threshold Exceeded? n
                Passphrase:
              DTMF over IP: out of band       Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? y
                                              Interworking Message: PROGress
```

2. Type *y* in the **Media Encryption?** field to enable Media Encryption on trunk calls using this signaling group.

   Note:

   Leaving this field in the default state (**n**) overrides the encryption administration on the IP Codec Set form (Media Encryption field on the IP Codec Set form on page 150) for any trunk call using this signaling group. That is, if the IP codec set that is used between two networks is administered as **aes** or **aea** (Table 22: Media Encryption Field Values (IP Codec Set) on page 151), then a call between two endpoints over a H.323 trunk using this IP codec set fails because there is no voice path.

   This field does not display if the **Media Encryption Over IP?** field is *n* on the **Customer Options** form (Media encryption field on Optional Features form on page 149).

3. Type an 8- to 30-character string in the **Passphrase** field.

   This string:

   - Must contain at least 1 alphabetic and 1 numeric symbol
   - Can include letters, numerals, and!&*?;'^(),.:-
   - Is case-sensitive

   You must administer *the same passphrase* on both signaling group forms at each end of the IP trunk connection. For example, if you have two systems A and B with trunk A-B between them, you must administer both Signaling Group forms with *exactly the same passphrase* for the A-to-B trunk connection.

   If you have previously administered a passphrase, a single asterisk (*) appears in this field. If you have not administered a passphrase, the field is blank.

   Note:

   This field does not display if either the:

   - **Media Encryption Over IP?** field on the **Customer Options** form (Media encryption field on Optional Features form on page 149) is *n*.

     or

   - **Media Encryption?** field on the **Signaling Group** form (Media encryption and passphrase fields for signaling groups on page 152) is *n*.

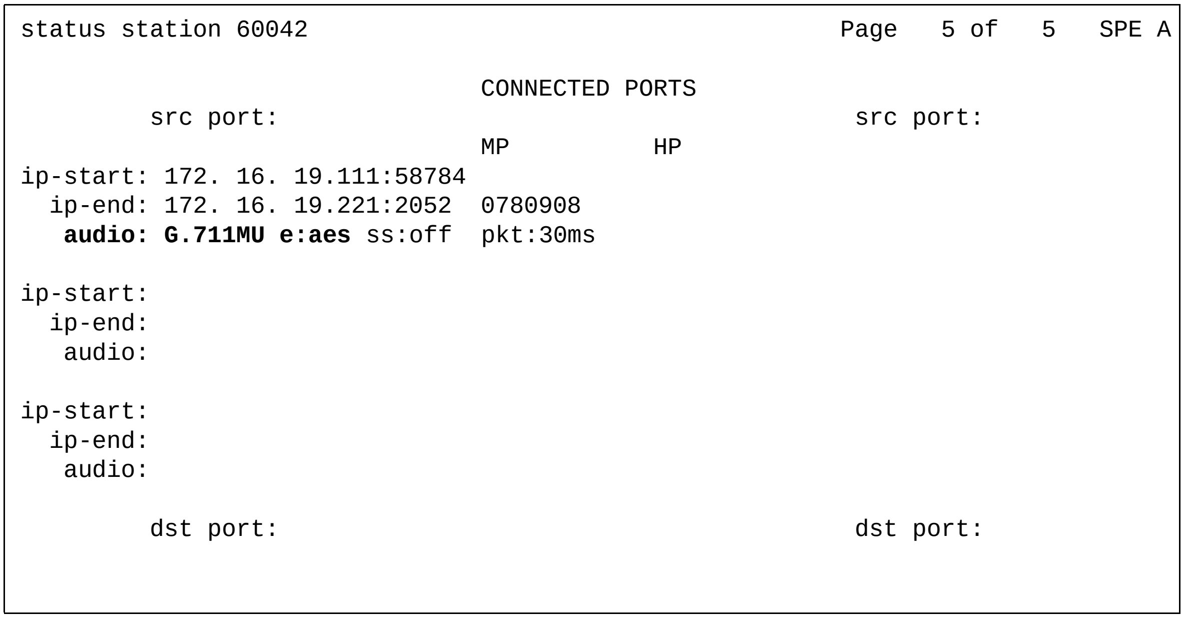
Viewing encryption status for stations and trunks

The current status of encryption usage by stations and trunks can be viewed using the **`status station`** and **`status trunk`** commands.

To check media encryption usage for a station:

1. Type **`status station <`*`extension`*`>`**, and go to the **Connected Ports** page.

Connected ports screen



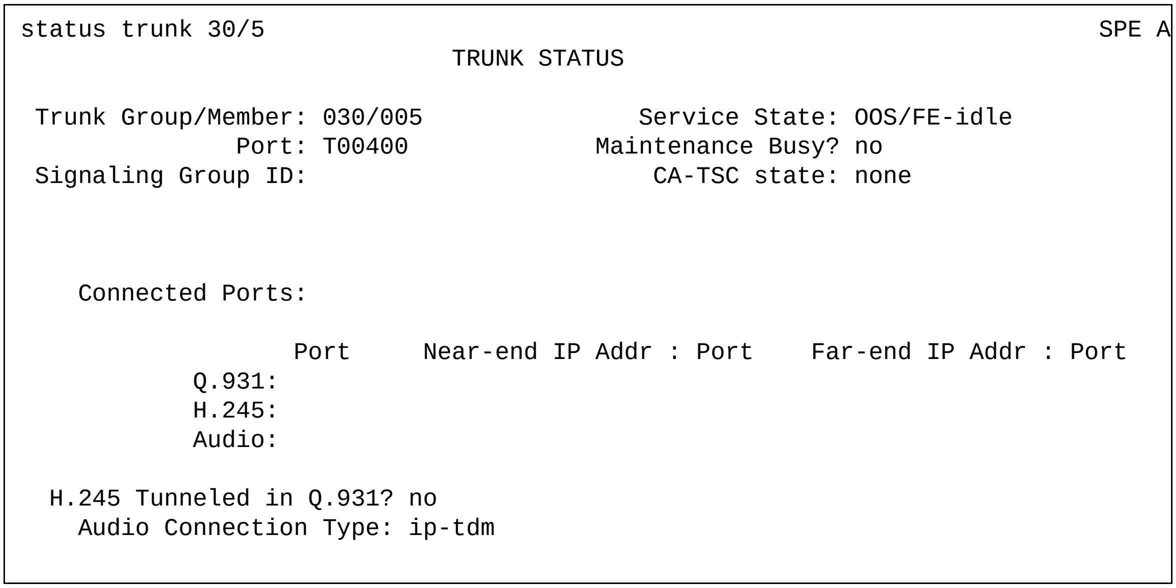
```
status station 60042                                     Page   5 of   5   SPE A

                                CONNECTED PORTS
         src port:                                        src port:
                                MP          HP
ip-start: 172. 16. 19.111:58784
  ip-end: 172. 16. 19.221:2052  0780908
   audio: G.711MU e:aes ss:off  pkt:30ms

ip-start:
  ip-end:
   audio:

ip-start:
  ip-end:
   audio:

         dst port:                                        dst port:
```

This screen shows that a port is currently connected and using a G711 codec with AES media encryption.

To check media encryption usage for a trunk:

1. Type **`status trunk <`*`group/member`*`>`**.

Media encryption status for a trunk group 30, member 5



```
status trunk 30/5                                                          SPE A
                              TRUNK STATUS

 Trunk Group/Member: 030/005               Service State: OOS/FE-idle
               Port: T00400             Maintenance Busy? no
 Signaling Group ID:                        CA-TSC state: none



    Connected Ports:

                   Port     Near-end IP Addr : Port    Far-end IP Addr : Port
            Q.931:
            H.245:
            Audio:

  H.245 Tunneled in Q.931? no
    Audio Connection Type: ip-tdm
```

This screen shows that trunk 5 is currently using no media encryption.

About legal wiretapping

If you receive a court order requiring you to provide law enforcement access to certain calls placed to or from an IP endpoint, you can administer Service Observing permissions to a selected target endpoint (see Service Observing in Table 24: Media Encryption interactions on page 156). Place the observer and the target endpoint in a unique Class of Restriction (COR) with *exactly the same properties and calling permissions* as the original COR, otherwise the target user might be aware of the change.

About possible failure conditions

Using Media Encryption in combination with an administered security policy might lead to blocked calls or call reconfigurations because of restricted media capabilities. For example, if the IP codec set that is used between two network regions is administered as **aes** and/or **aea** (Table 22: Media Encryption Field Values (IP Codec Set) on page 151), then if a call between two endpoints that do not support at least one common encryption option (one in each region) is set up, there is no voice path.

How does Media Encryption interact with other features?

Media Encryption does not affect most Communication Manager features or adjuncts, except for those listed in Table 24: Media Encryption interactions on page 156.

Table 24: Media Encryption interactions

Interaction	Description
Service Observing	You can Service Observe a conversation between encrypted endpoints. The conversation remains encrypted to all outside parties except the communicants and the observer.
Voice Messaging	Any call from an encryption-enabled endpoint is decrypted before it is sent to a voice messaging system. When the TN2302AP IP Media Processor circuit pack receives the encrypted voice stream, it decrypts the packets before sending them to the voice messaging system, which then stores the packets in unencrypted mode.
Hairpinning	Hairpinning is not supported when one or both media streams are encrypted, and Avaya Communication Manager does not request hairpinning on these encrypted connections.
VPN	Media encryption complements virtual private network (VPN) security mechanisms. Encrypted voice packets can pass through VPN tunnels, essentially double-encrypting the conversation for the VPN “leg” of the call path.
H.323 trunks	Media Encryption behavior on a call varies based on these conditions at call set up: ● Whether shuffled audio connections are permitted ● Whether the call is an inter-region call ● Whether IP trunk calling is encrypted or not ● Whether the IP endpoint supports encryption ● The media encryption setting for the affected IP codec sets These conditions also affect the codec set that is available for negotiation each time a call is set up.

About network management

Network management is the practice of using specialized software tools to monitor and maintain network components. Proper network management is a key component to the high availability of data networks.

The two basic network management models are:

- Distributed. Specialized, nonintegrated tools (and sometimes organizations) to manage discrete components
- Centralized. Integrating network management tools and organizations for a more coherent management strategy.

For a detailed discussion of Avaya's network management products, common third-party tools, and the distributed and centralized management models, see *Avaya Application Solutions: IP Telephony Deployment Guide*, 555-245-600.

This section touches briefly on the following topics:

- About link recovery
- Controlling QoS policies
- Monitoring network performance

About link recovery

The H.248 link between a media server running Avaya Communication Manager and a media gateway, and the H.323 link between a media gateway and an H.323-compliant IP endpoint, provide the signaling protocol for

- Call setup
- Call control (user actions such as Hold, Conference, or Transfer) while the call is in progress
- Call tear-down

If the link goes down, Link Recovery preserves any existing calls and attempts to re-establish the original link. If the gateway/endpoint cannot reconnect to the original server/gateway, then Link Recovery automatically attempts to connect with alternate TN799DP (C-LAN) circuit packs within the original server's configuration or to a Local Survivable Processor (LSP).

See *Maintenance Procedures for Avaya Communication Manager 2.2, Media Gateways and Servers, (03-300192)*, for details of the link recovery process and administration options.

Controlling QoS policies

Avaya Policy Manager is a network management tool that allows you to control Quality of Service (QoS) policies in your IP voice network consistently:

- Avaya Policy Manager helps you implement QoS policies consistently for both the data and the voice networks.
- QoS policies are assigned according to network regions and are distributed through the Enterprise Directory Gateway to your systems and to routers and switching devices.

Figure 10: Avaya Policy Manager application sequence on page 158 illustrates how Avaya Policy Manager works.

Figure 10: Avaya Policy Manager application sequence

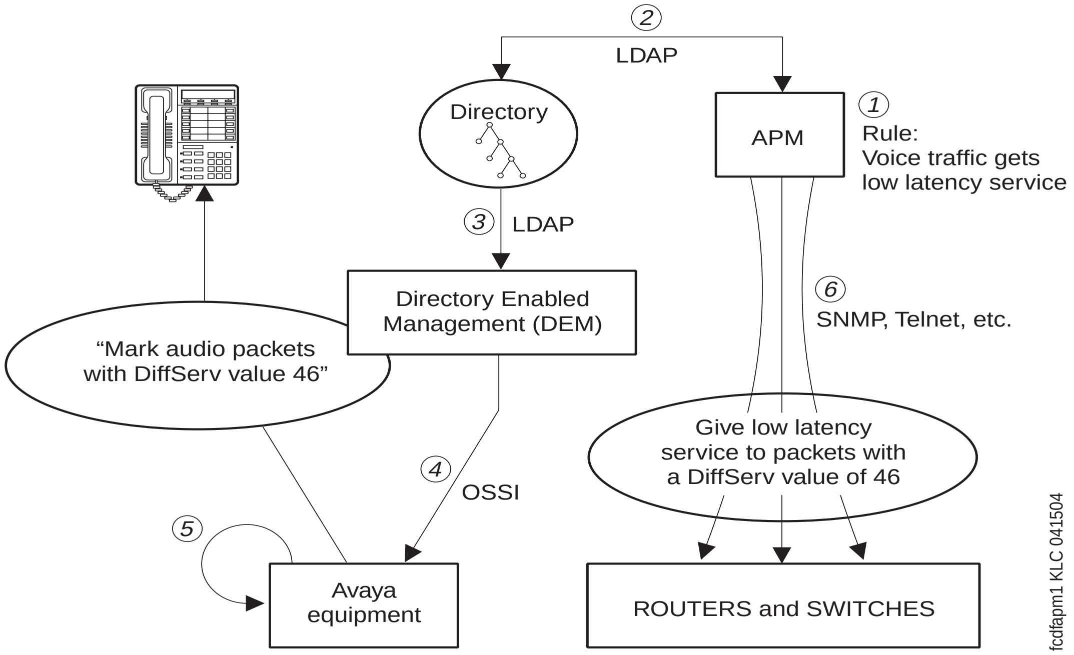


Figure notes:

1. **Business rule established in Avaya Policy Manager**
2. **Avaya Policy Manager uses LDAP to update Communication Manager**
3. **Directory Enabled Management (DEM) identifies the change in the directory.**
4. **EDG updates Communication Manager administration through the Ethernet switch**
5. **Communication Manager tells the Media Processor, CLAN, and IP Phones to mark audio packets with DSCP=46.**
6. **Avaya Policy Manager distributes policy information to other network devices, including low latency service for DiffServ value of 46.**

For more information about Avaya Policy Manager, see your Avaya representative.

Monitoring network performance

The Avaya VoIP Monitoring Manager, a VoIP Network Quality monitoring tool, allows you to monitor these quality-affecting network factors:

- Jitter levels
- Packet loss
- Delay
- CODECs used
- RSVP status

For more information about Avaya VoIP Monitoring Manager, see *Avaya Application Solutions: IP Telephony Deployment Guide*, 555-245-600.

Chapter 4: Administering dedicated networks

This chapter contains these main sections:

- Distributed Communications System contains a description of DCS and the features that can be used transparently on a DCS network. This section also contains a description of some pre-requisite DCS features:
  - Prerequisite DCS administration
  - DCS signaling
  - Gateway switch
  - Italian DCS Protocol
  - DCS configurations with AUDIX
- QSIG contains a description of QSIG and how to administer it.
  - Overview
  - QSIG/DCS interworking
  - Offer level functionality
  - Basic call setup
  - Transfer into QSIG Message Center
  - Value-Added (VALU) MSI
  - QSIG Centralized Attendant Services (CAS)
  - Call-independent Signaling Connection (CISCs)
  - About Non-Call Associated Temporary Signaling Connection (NCA-TSC)
  - Migrating to QSIG: some considerations
- Centralized Attendant Service contains a description of CAS and features, considerations, and feature interactions.
  - What is Centralized Attendant Service (CAS)
  - Administering CAS
- Extended Trunk Access contains description, administration, and feature interactions for ETA.
  - What is Extended Trunk Access (ETA)
  - Administering Extended Trunk Access

- Inter-PBX Attendant Service contains description, administration, and feature interactions for Inter-PBX Attendant Service.
  - What is Inter-PBX Attendant Service (IAS)
  - Administering Inter-PBX Attendant Service
  - About Inter-PBX Attendant Service interactions
- ISDN Feature Plus is a description of ISDN Plus networking capabilities.
  - What is ISDN Feature Plus
  - Administering ISDN Feature Plus
  - Differences in Inserted Digits field
  - About interrogation between message center and served user switches
- Centralized Voice Mail describes how to administer Centralized Voice Mail with Mode codes.
  - About centralized voice mail
  - What are mode code centralized voice mail configuration requirements
  - Administering Centralized Voice Mail
- Japan TTC Q931-a is a brief description of Japan TTC private networking protocols.
  - About Japan TTC Q931-a
  - Considerations about TTC Basic Call Setup with Number Identification Supplementary Service
  - What are the TTC Q931-a Protocols
  - Administering Japan TTC Q931-a

Note:

> See Chapter 5: Feature interactions and considerations for feature interaction information and other considerations for using the features described in this chapter.

Distributed Communications System

Distributed Communications System (DCS) allows you to configure 2 or more switches as if they were a single, large switch. DCS provides attendant and voice-terminal features between these switch locations. DCS simplifies dialing procedures and allows transparent use of some of the Communication Manager features. (Feature transparency means that features are available to all users on DCS regardless of the switch location.)

Table 25: DCS topics, descriptions, and administration links on page 163 give you links to the main DCS topics, descriptions, and to administration procedures.

Table 25: DCS topics, descriptions, and administration links ***1 of 2***

Topic	Description	Administration
Prerequisite DCS administration	● Uniform Dial Plan	● In order to configure a network using DCS, Uniform Dial Plan (UDP) must be administered on your systems. For more information on UDP administration see the *Administrator's Guide for Avaya Communication Manager (555-233-506).*
	Private Network Access	● Administering Private Network Access
	● Extension Number Portability	● Administering Extension Number Portability
		1 of 2

Table 25: DCS topics, descriptions, and administration links ***2 of 2***

Topic	Description	Administration
DCS signaling	● DCS over ISDN-PRI D-channel (DCS+) ● Asynchronous PPP over analog trunks ● ISDN/X.25 gateway ● Italian DCS Protocol	● Administering DCS+ over ISDN-PRI D-channel ● Administering Italian DCS (Enhanced DCS)
DCS configurations with AUDIX		● Administering a 2-node private network with AUDIX ● Administering a 3-node public/private network with AUDIX
		2 of 2

Prerequisite DCS administration

Before administering Communication Manager for DCS you must first complete these administration procedures:

- Uniform Dial Plan
- Private Network Access
- Extension Number Portability

Uniform Dial Plan

In order to configure a network using DCS, Uniform Dial Plan (UDP) must be administered on your systems. For more information on UDP administration see the *Administrator's Guide for Avaya Communication Manager (555-233-506).*

Private Network Access

Use Private Network Access to allow calls to other switching systems in a private network. These calls do not use the public network. They are routed over customer-dedicated facilities.

Administering Private Network Access

To administer Private Network Access fill out the forms and fields as indicated in Table 26: Private Network Access administration on page 165.

Table 26: Private Network Access administration

Screen	Field
Trunk Groups Access **APLT** **ISDN-BRI** **ISDN-PRI** Tandem	● **Group Type** field is **access**, **aplt**, **tandem**, **tie**, or **isdn**. ● **Service Type** field is **access**, **tie**, or **tandem**. ● Complete **COR** digit treatment and common type fields for tie trunk groups associated with a private network.
Class of Restriction	● Advanced Private Line Termination
Feature Access Code (FAC)	● Automatic Alternate Routing Access Code
Dialplan Analysis form	● All
AAR and ARS Digit Conversion Table	● All
Node Number Routing	● All
Station	● **COR**

Note:

Private networks can include:

- Common-control switching arrangement (**CCSA**)
  - Unless prohibited by the COR, all incoming private network trunks, except CCSA, can access outgoing trunks without attendant or terminal-user assistance. All incoming CCSA calls must route to an attendant or a terminal user.
  - When off-network calling is part of the **CCSA** and Enhanced private-switched communications service (**EPSCS)**, long-distance calls route as far as possible over these networks before terminating on the public network. Thus, charges for toll calls are reduced. The COR you administer to individual system users determines whether access to this capability is allowed or denied.
- Distributed Communications Systems (**DCS**) and Enhanced DCS (**EDCS**)

- Electronic tandem network (**ETN**)
- Enhanced private-switched communications service (**EPSCS**)
- Tandem-tie-trunk network (**TTTN**)
- Italian Traslatore Giunzione Uscente/Entrante/Interno (**TGU**/**TGE**/**TGI)** trunks. These trunks provide private network access between 2 switching systems. They also provide some feature transparency for COR (Inward Restriction), DID (when reaching busy stations), and Intrusion.
- QSIG Trunks (for more information, see Overview on page 186)
- IP Trunks

Extension Number Portability

The ENP Numbering Plan allows you to set 4- or 5-digit extensions in the ENP subnetwork to a 7-digit AAR-like number that is sent to other nodes in the network. Only the first 1 or 2 leading digits of the extension are significant. ENP Codes are distinguished from AAR location codes because ENP Codes are home on every node within the ENP subnetwork, and ENP Codes are administered in the ENP Numbering Plan table as well as in the AAR Analysis table. Since ENP Codes are home on every node, they cannot be used as AAR location codes.

UDP extensions are converted to ENP numbers if node number routing is specified for the extensions in the UDP table.

Note:

> One ENP code is required for a 4-digit ENP subnetwork. A 5-digit UDP requires one ENP code for each leading digit of extensions used within the ENP subnetwork.

DCS message signaling links are not required to support ENP. As a result, many multiple switch configurations are possible with ENP. Typically the ENP network will be a subnetwork of a UDP or Electronic Tandem Network (ETN).

Administering Extension Number Portability

To administer ENP fill out the forms and fields as indicated in Table 27: Extension Number Portability administration on page 167.

Table 27: Extension Number Portability administration

Screen	Field
AAR and ARS Digit Conversion Tables	● Assign all 3-digit **ENP** codes as home, and if using a 5-digit **UDP**, associate the **ENP** codes with the leading, or 10 thousands, digit (that is, the fifth digit of the extension). For example, for extension number 73446, “7” is the 10 thousands digit.
Extension Number Portability Numbering Plan	● Associate the leading one or two digits of extensions in the **ENP** subnetwork with a 3-digit **ENP** code, used to construct a 7-digit **AAR**-like **ENP** number.
Node Number Routing	● Associate a route pattern with each node in the **ENP** subnetwork.
Uniform Dialing Plan	● **Ext Code** field: Enter the number of digits in the plan (4 or 5) and the Extension Codes for non-home extensions in the **ENP** subnetwork as ENPNode (node number routed).

DCS signaling

In addition to tie-trunk connections for the transmission of voice and call-control data, DCS requires a special signaling connection to carry the information needed to make the DCS features work.

This signaling connection, or link, between two switches in a DCS network can be implemented in one of four ways:

- Over an Integrated Services Digital Network (ISDN) - Primary Rate Interface (PRI) signaling (or “D”) channel (see DCS over ISDN-PRI D-channel (DCS+)).
- Over Asynchronous PPP over analog trunks
- Over ISDN/X.25 gateway
- Over a TCP/IP— either
  - point-to-point protocol (PPP) connection

    or
  - 10/100Base-T Ethernet connection

Gateway switch

When a DCS network uses a mixture of two or three of the different DCS signaling types, one or more switches in the network must act as a gateway. A gateway switch is connected between two switches using different signaling protocols, enabling the two end switches to communicate by converting the signaling messages between the two protocols. A gateway switch can provide conversion between two or all three of the signaling protocols, but only one protocol can be used for DCS signaling between any two switches.

DCS over ISDN-PRI D-channel (DCS+)

AT&T SDN, as well as MCI N-Quest Service provide for the transmission of the DCS protocol across the public network, as a virtual private network. DCS Over ISDN-PRI D-channel (DCS+) permits access to the public network for DCS connectivity between DCS switch nodes.

ISDN B-channel (bearer)

ISDN is a widely-adopted means of private networking that uses the 24-channel ISDN-PRI trunk with a bandwidth of 1.544 Mb/sec. Channels 1-23 are used for voice (bearer) data, digitized conversations that can be interleaved among the 23 bearer channels.

ISDN D-channel (signaling)

Channel 24 is reserved for call signaling messages that perform basic call set-up, maintenance, and tear-down as well as DCS+ or QSIG messages that can seamlessly integrate two switches in different locations of an enterprise network.

MA-UUI

DCS Over ISDN-PRI utilizes the Message-Associated User-to-User Information (MA-UUI) and Temporary Signaling Connections (TSC) to transport certain DCS control information (see Temporary signaling connections (TSC)). MA-UUI allows additional user-specific information to be transported along with certain ISDN call-control messages.

Note:

Use this feature only over DS1/E1 or T1 circuit packs that are administered to Country Protocol Option 1, Protocol Version A (even in a private network environment) independent of what country the system is in.

DCS+ configurations

DCS+ network configurations can be:

- TCP/IP DCS network — A DCS network configured with 2 or more switches using TCP/IP (PPP or 10/100BaseT Ethernet) signaling for transporting DCS feature transparency information.
- Traditional DCS network — A DCS network configured with 2 or more switches using BX.25 signaling for transporting DCS feature transparency information.
- D-channel DCS network (private network only) — A DCS network that includes a switch using the ISDN-PRI D-channel DCS transparency information (D-channel signaling). ISDN-PRI facilities with this type of network use only private-line facilities.
- D-channel DCS network (public network access/egress) — A DCS network that includes a switch using D-channel signaling. At least one of these ISDN-PRI facilities uses a public network ISDN-PRI.
- Integrated DCS network (private network only) — A DCS network that contains a variety of switches using TCP/IP, BX.25, or D-channel signaling methods. At least one Avaya switch serves as an ISDN-PRI DCS Gateway node. This node can interwork DCS transparency information between the three signaling protocols.

  An ISDN-PRI DCS Gateway node provides backward compatibility to existing traditional DCS networks.
- Integrated DCS network (public network access) — The same as D-channel DCS Network (Private Network Only), but the D-channel of at least one ISDN- PRI facility uses a public network ISDN-PRI.

For more information on DCS+ configurations use these links to the *ATAC 2003 Connectivity Guide*:

- **Associates**: http://associate2.avaya.com/sales_market/tools/proposal-create/CG/
- **Business Partners**: https://www.avaya.com/doc/gpp/public/?_requestid=102567 (does not include a Chapter on UPS Power)
- **All others**: http://www.avayaups.com/ (Website outside the Avaya firewall and available to customers)

Temporary signaling connections (TSC)

A TSC provides a temporary signaling path through ISDN switches for exchanging supplementary service information on ISDN-PRI D-channels. There is no B-channel related to the connection; no data or voice transmissions take place.

There are two types of temporary signaling connections:

- Call Associated TSC
- Non-Call Associated (NCA-TSC)

Call Associated TSC

Call Associated TSC (CA-TSC) refers to a service for exchanging USER INFORMATION messages associated with an ISDN B-channel connection by the call reference value of the call control data packets. On an Avaya switch, this type of TSC is used only for DCS features on ISDN-PRI Signaling Groups administered with Supplementary Service Protocol **a**.

Non-Call Associated (NCA-TSC)

An NCA-TSC is a connection not related with any ISDN B-channel connections. Communication Manager supports two types of NCA-TSC that conform to two different protocol standards:

- The AT&T type of NCA-TSC is used for the DCS Over ISDN-PRI D-channel and DCS AUDIX applications. Only ISDN-PRI Signaling Groups administered with Supplementary Service Protocol **a** support AT&T NCA-TSCs.

  An AT&T NCA-TSC is an administered virtual connection established for exchanging USER INFORMATION messages on the ISDN D-channel. Once an AT&T NCA-TSC has been administered and enabled, it is active for an extended period of time. There are two types of administered NCA-TSCs depending on their setup mechanism:

  - Permanent (can be established by near-end or far-end)
  - As-needed

  Once enabled, a permanent NCA-TSC remains established while the system is running. If the permanent NCA-TSC drops for any reason, the system attempts to reestablish the connection. An as-needed administered NCA-TSC is established based on user request and the availability of TSC facilities. The connection drops after an administered period of inactivity.

The system can transport DCS or DCS AUDIX messages over an ISDN-PRI D-channel and over BX.25 data links when functioning as a gateway between a switch equipped with DCS Over ISDN-PRI D-channel and a switch equipped with traditional DCS using BX.25 data links. In this situation, the messages travel from the gateway through the NCA-TSCs or CA-TSCs to TSC-capable switches and from the gateway to switches that support only traditional DCS using a BX.25 logical channel.

At least one switch must be configured as an ISDN DCS Gateway node in a DCS network that consists of switches that support DCS Over ISDN-PRI D-channel and PBXs that do not support the feature. Switches directly connected to AUDIX systems serve as Gateway nodes.

Administering DCS+ over ISDN-PRI D-channel

To administer DCS+ fill out the forms and fields as indicated in Table 28: DCS+ administration on page 171.

Table 28: DCS+ administration ***1 of 2***

Screen	Field
Signaling Group	Page 1: ● Max number of **NCA TSC** ● Max number of **CA TSC** ● Trunk Group for **NCA TSC** Page 2: ● Administered **NCA TSC** Assignment fields ● Service/Feature ● Inactivity Time-out (min)
ISDN TSC Gateway Channel Assignments	● All
Trunk Group (**ISDN-PRI**)	Page 2: ● Used for **DCS** Node Number **DCS** Signaling ● **NCA TSC** Trunk Member
Route Pattern	Page 1: ● **TSC** ● **CA TSC** Request
	1 of 2

Table 28: DCS+ administration ***2 of 2***

Screen	Field
Processor Channel Assignment	● Application
Feature-Related System Parameters	● Record **NCA-TSC**s for **CDR**
CDR System Parameters	Page 1: ● Record Non-Call-Assoc **TSC**? ● Record Call-Assoc **TSC**?
	2 of 2

Note:

There are several differences in administration between switches. For example, PRI is translated a little differently in Avaya DEFINITY G3r when traditional DCS and this feature are used in combination. On systems with AUDIX in a DCS environment, an additional column has been added to the Signaling Group screen so you can specify which AUDIX system and switch to use. When traditional DCS and DCS+ (over ISDN D-channel) are used in combination, translations are also different.

Asynchronous PPP over analog trunks

Asynchronous linking also provides the capability of DCS connectivity over analog trunks. A router and an external modem help provide this capability. The router converts the Ethernet IP packets to be transmitted over analog facilities using PPP using the external modem.

ISDN/X.25 gateway

An Avaya switch can serve as an interface between switches that support the D-channel signaling feature and those that do not support this feature. The switch providing this interface is known as the ISDN-DCS Gateway node and provides backward compatibility to existing traditional DCS networks.

The ISDN-DCS Gateway node maintains a mapping between processor channels and administered NCA-TSCs. When a DCS D-channel message arrives on an Administered NCA-TSC acting as a gateway, it is converted to a traditional DCS message and sent out through the processor channel that has been administered to map to this administered NCA-TSC. Likewise, when a traditional DCS message arrives at the gateway node on a processor channel acting as a gateway, it is converted to a DCS D-channel message and sent out through the administered NCA-TSC that has been associated with this processor channel on the ISDN Gateway Channel screen.

A gateway is required whenever a transition is being made from BX.25 signaling to D-channel signaling. When the transition takes place at a switch that sits between that part of the network that supports D-channel DCS and that part that does not, that switch is an ISDN-DCS Gateway. A DCS network consisting entirely of switches that support D-channel DCS never requires an ISDN-DCS Gateway because none of the switches require “translation” to/from BX.25.

Italian DCS Protocol

Italian DCS Protocol (also known as Enhanced DCS) adds features to the existing DCS capabilities. EDCS is used primarily in Italy. EDCS adds the following features:

- Exchanging information to provide class of restriction (COR) checking between switches in the EDCS network
- Providing call-progress information for the attendant
- Allowing attendant intrusion between a main and a satellite
- Allowing a main switch to provide DID/CO intercept treatment rather than the satellite switch.

Note:

EDCS is not compatible with DCS Over/Under ISDN-PRI. With EDCS, all nodes must use EDCS. If used with ISDN-PRI, configure the switch as a DCS node. Also, DCS-ISDN display enhancements are not currently available in EDCS.

Administering Italian DCS (Enhanced DCS)

Screen	Field
Feature-Related System-Parameters	Page 1: ● ITALIAN DCS PROTOCOL **Italian Protocol Enabled?** *y* ● **Apply Intercept Locally?** ● **Enforce PNT-to-PNT Restrictions~~?~~**

DCS configurations with AUDIX

The following two examples provide details for setting up two basic DCS networks:

- Administering a 2-node private network with AUDIX
- Administering a 3-node public/private network with AUDIX

The first is a two-node network and the second is a three-node network. These examples use BX.25 and D-Channel signaling connections.

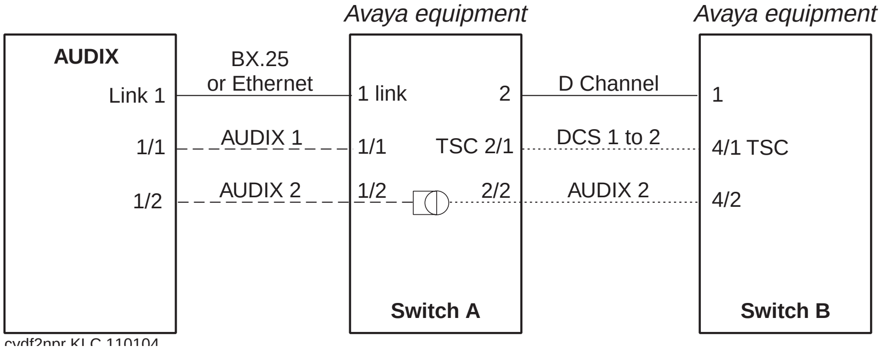
Administering a 2-node private network with AUDIX

2-Node private network on page 174 shows a 2-node DCS/AUDIX D-channel network. In this configuration, DCS feature transparency is achieved exclusively through the exchange of user-to-user information on the D-channel using one of the three methods discussed earlier — MA-UUI, CA-TSCs or NCA-TSCs. Although NCA-TSCs are nothing more than virtual connections on the D-channel, they are shown as independent entities in the diagram for the purposes of clarity. Administered TSC 2/1 (that is, the first Administered NCA-TSC of signaling group 2) of Switch A is connected to TSC 4/1 of Switch B. This virtual connection is used in the exchange of user-to-user information for DCS features not associated with any current B-channel connection.

Notice that for AUDIX, a BX.25 data link is no longer required between the host switch and the remote switch(es). AUDIX messages between the AUDIX system and the remote switch will use the AUDIX Gateway functionality of the host switch and will be transported to the remote switch using an NCA-TSC. Specifically, AUDIX messages destined for Switch B will arrive at Switch A on Link 1, Channel 2 (processor channel 57), be converted to ISDN-PRI Q.931 format and sent out using Administered NCA-TSC 2/2.

This is accomplished by administering processor channel 57 as a gateway and mapping it on the gateway screen to Administered NCA-TSC 2 of signaling group 2 that is also administered as a gateway.

Figure 11: 2-Node private network



The following tables show you how you would complete each of the necessary screens.

AUDIX administration

- **AUDIX Translations** screen

Switch Number	AUDIX Port	Switch Port[1]	Logical Channel	Data Link
1	1	59	1	1
2	2	57	2	1

[1] Switch Port refers to the processor channel that is used for AUDIX in the switch.

Communication Manager administration (switch 1)

- **Dial Plan Analysis** screen

Dialed String	Total Length	Call Type
4	4	ext
5	4	ext

- **Uniform Dial Plan** screen

Matching Pattern	Len	Del	Insert Digits	Net	Node Num
5	4	**0**	222	aar	2
6	4	0	223	aar	2

- **AAR Digit Conversion** screen

Matching Pattern	Min	Max	Del	Replacement String	Net	Conv	ANI Req
221	7	7	3	-	ext	n	

- **AAR Digit Analysis Report**

Dialed String	Min	Max	Rte Pattern	Call Type	Node Num
222	7	7	2	aar	2

- **Signaling Group** screen (signaling group 2)

TSC	**Local**	**Enabled**	**Establish**	**Dest.**	**Far-end**	**Appl.**
Index	Ext.			Ext.	**Switch**-ID	
1	4900	y	permanent	5900	2	dcs
2	4901	y	permanent	5901	-	gateway

- **Trunk Group** screen

Group #	**Grp Type**	**Used for DCS**	**DCS Sig. Method**	**Switch ID**
2	isdn-pri	y	d-chan	2

- **Routing Pattern** screen

Routing Pattern #	**Trunk Group #**	**FRL**	**Del**	**TSC**	**CA-TSC Request**
2	2	0	3	y	at-setup

- **Gateway Channel** screen

Signaling Group	**TSC Index**	**Processor Channel**	**Application**
2	2	57	audix

- **Processor Channel** screen

Proc Channel	**Application**	**Inter. Link**	**Channel**	**Remote Proc. Channel**	**Switch ID**
57	gateway	1	2	2	-
59	audix	1	1	1	1

Communication Manager administration (switch 2)

- **Dial Plan Analysis** screen

Dialed String	**Total Length**	**Call Type**
4	4	ext
5	4	ext

- **Uniform Dial Plan** screen

Matching Pattern	**Len**	**Del**	**Insert Digits**	**Net**
4	4	**0**	221	aar

- **AAR Digit Conversion** screen

Matching Pattern	**Min**	**Max**	**Del**	**Replacement String**	**Net**	**Conv**
222	7	7	3	-	ext	n

- **AAR Analysis Table**

Dialed String	**Min**	**Max**	**Rte Pat**	**Call Type**	**Node Num**
221	7	7	1	aar	1

- **Signaling Group** screen (signaling group 4)

TSC Index	**Local Ext.**	**Enabled**	**Establish**	**Dest.**	**Far-end Ext.**	**Appl Switch-ID**
1	5900	y	permanent	4900	1	dcs
2	5901	y	permanent	4901	-	audix

- **Trunk Group** screen

Group #	Grp Type	Used for DCS?	DCS Sig. Method	PBX ID
1	isdn-pri	y	d-chan	1

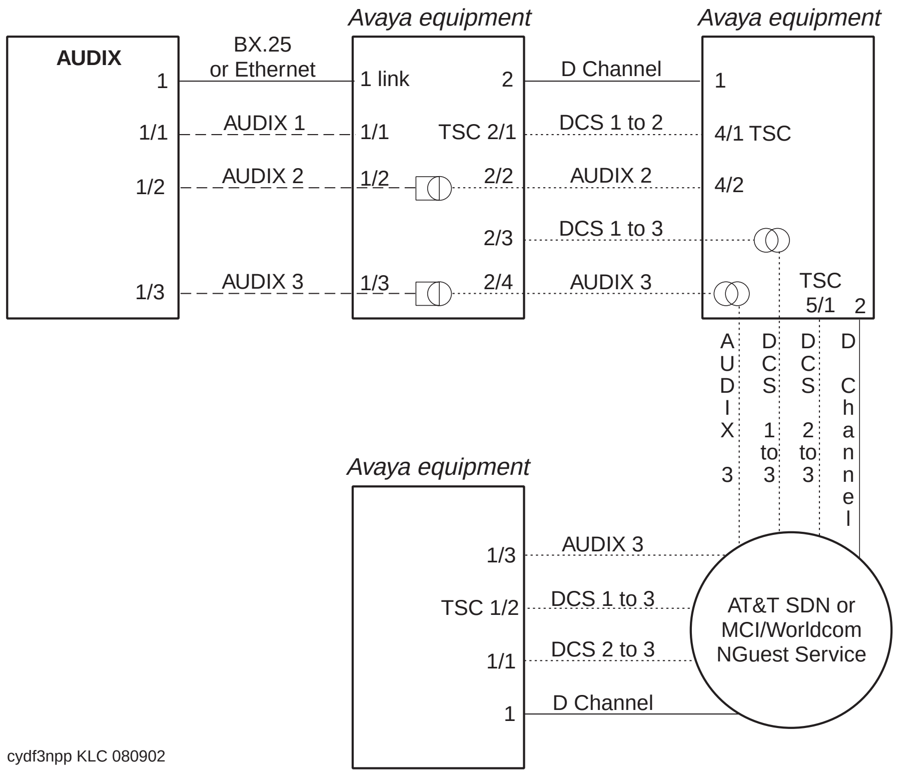
- **Routing Pattern** screen

Routing Pattern #	Trunk Group #	FRL	Del	TSC	CA-TSC Request
1	1	0	3	y	at-setup

Administering a 3-node public/private network with AUDIX

The D-channel signaling feature expands the domain of DCS networks by supporting configurations that include public network ISDN facilities utilizing network services including Software Defined Network (SDN). By eliminating the need for dedicated private line facilities, this feature allows geographically dispersed DCS networks to be cost effective. Figure 12: 3-Node public/private network on page 179 shows a 3-node network.

Figure 12: 3-Node public/private network



The following tables show you how you would complete each of the necessary screens.

AUDIX administration

- **AUDIX Translations** screen

Switch Number	AUDIX Port	Switch Port[1]	Logical Channel	Data Link
1	1	59	1	1
2	2	57	2	1
3	3	58	3	1

[1] Switch Port refers to the processor channel that is used for AUDIX in the PBX.

Communication Manager administration (switch 1)

- **Dial Plan Analysis Table**

Dialed String	Total Length	Call Type
4	4	ext
5	4	ext
6	4	ext

- **Uniform Dial Plan Table**

Ext Code	Type	Location Code
5xxx	**UDP**code	222
6xxx	**UDP**code	223

- **AAR Digit Conversion** screen

Matching Pattern	Min	Max	Del	Replacement String	Net	Conv
221	7	7	3	-	ext	n

- **AAR Analysis Table**

Dialed String	Min	Max	Rte Pat	Call Type	Node Num
222	7	7	2	aar	2
223	7	7	3	aar	3

- **Signaling Group** screen (signaling group 2)

TSC Index	Local Ext.	Enabled	Establish	Dest. Ext.	Far-end PBX-ID	Appl
1	4900	y	permanent	5900	2	dcs
2	4901	y	permanent	5901	-	gateway
3	4902	y	permanent	6902	3	dcs
4	4903	y	permanent	6903	-	gateway

- **Trunk Group** screen

Group #	Grp Type	Used for DCS?	DCS Sig. Method	PBX ID
2	isdn-pri	y	d-chan	2
3	isdn-pri	y	d-chan	3

- **Routing Pattern** screen

Routing Pattern #	Trunk Group #	FRL	Del	TSC	CA-TSC Request
2	2	0	3	y	at-setup
3	3	0	3	y	at-setup

- **Gateway Channel** screen

Signaling Group	TSC Index	Processor Channel	Application
2	2	57	audix
2	4	58	audix

- **Processor Channel** screen

Proc Channel	Application	Inter. Link	Channel	Remote Proc. Channel	PBX ID
59	audix	1	1	1	1
57	gateway	1	2	2	-
58	gateway	1	3	3	-

Communication Manager administration (switch 2)

- **Dial Plan Analysis Table**

Dialed String	Total Length	Call Type
4	4	ext
5	4	ext
6	4	ext

- **Uniform Dial Plan** screen

Ext Code	Type	Location Code
4xxx	**UDP**code	221
6xxx	**UDP**code	223

- **AAR Digit Conversion** screen

Matching Pattern	Min	Max	Del	Replacement String	Net	Conv
222	7	7	3	-	ext	n

- **AAR Analysis Table**

Dialed String	Min	Max	Rte Pat	Call Type	Node Num
221	7	7	1	aar	1
223	7	7	3	aar	3

- **Signaling Group** screen

  Signaling group 4

TSC	Local	Enabled	Establish	Dest.	Far-end	Appl.
Index	Ext.			Ext.	**PBX**-ID	
1	5900	y	permanent	4900	1	dcs
2	5901	y	permanent	4901	-	audix

  Signaling group 5

TSC	Local	Enabled	Establish	Dest.	Far-end	Appl.
Index	Ext.			Ext.	**PBX**-ID	
1	5905	y	permanent	6905	3	dcs

- **Trunk Group** screen

Group #	Grp Type	Used for DCS?	DCS Sig. Method	PBX ID	NCA-TSC Sig. Group[1]
1	isdn-pri	y	d-chan	1	-
3	isdn-pri	y	d-chan		5

[1] This field is only used for tandeming.

- **Routing Pattern** screen

Routing Pattern #	Trunk Group #	FRL	Del	TSC	CA-TSC Request
1	1	0	3	y	at-setup
3	3	0	3	y	at-setup

Communication Manager administration (switch 3)

- **Dial Plan Analysis Table**

Dialed String	Total Length	Call Type
4	4	ext
5	4	ext
6	4	ext

- **Uniform Dial Plan** screen

Ext Code	Type	Location Code
4xxx	**UDP**code	221
5xxx	**UDP**code	222

- **AAR Digit Conversion** screen

Matching Pattern	Min	Max	Del	Replacement String	Net	Conv
223	7	7	3	-	ext	n

- **AAR Analysis Table**

Dialed String	Min	Max	Rte Pat	Call Type	Node Num
221	7	7	1	aar	1
222	7	7	1	aar	2

- **Signaling Group screen** (signaling group 4)

TSC Index	Local Ext.	Enabled	Establish	Dest. Ext.	Far-end PBX-ID	Appl.
1	6905	y	permanent	5905	2	dcs
2	6902	y	permanent	4902	1	dcs
3	6903	y	permanent	4903	-	audix

- **Trunk Group** screen

Group #	Grp Type	Used for DCS?	DCS Sig. Method	PBX ID
1	isdn-pri	y	d-chan	

- **Routing Pattern** screen

Routing Pattern #	Trunk Group #	FRL	Del	TSC	CA-TSC Request
1	1	0	3[1]	y	at-setup

[1] Should be blank if SDN network routing requires 7 digits.

QSIG

The main QSIG topics in this section include

Note:

See Chapter 5: Feature interactions and considerations for feature interaction information and other considerations when using QSIG.

Overview

QSIG is the generic name for a family of signaling protocols. The Q reference point or interface is the logical point where signaling is passed between two peers in a private network. QSIG signaling allows certain features to work in a single-vendor or multi-vendor network.

In the mid-1980s the networking signaling protocols of PBX manufacturers’ were proprietary, causing problems for customers using several different PBX systems in their networks. Under the auspices of the ISDN Private Network Systems (IPNS) Forum, an initiative was started in Europe to create a standardized network signaling protocol to link dissimilar PBXs intelligently. The resulting signaling standard, QSIG, supports:

- Basic call set-up and tear-down
- Limited-digit dialing across PBXs through a common dial plan
- Sending and receiving telephone display information (for example calling name and number)
- Feature-transparency for a limited set of features

QSIG complies with the International Standardization Organization (ISO) for Integrated Services Digital Network (ISDN) private-networking specifications. QSIG is defined by ISO as the worldwide standard for private networks. QSIG uses ISO standard protocols as well as call-independent signaling connections (CISCs), administered as non-call-associated temporary signaling channels (NCA-TSCs).

QSIG/DCS interworking

In general, no features are interworking between QSIG and DCS, with the following exceptions (valid only with DCS+):

- Name and number transport
- Voicemail
- Leave word calling

Offer level functionality

Communication Manager provides different levels of QSIG functionality. You can view the status of each level on the **System Parameters Customer Options** screen.

Table 29: QSIG features supported by Avaya Communication Manager on page 187 lists the QSIG features supported by Communication Manager at each offer level. Valu-added (VALU) MSI is included in Supplementary Services, but is separated in the table, because the features that use Manufacturer Specific Information (MSI) only work between Avaya systems (see note below).

Table 29: QSIG features supported by Avaya Communication Manager ***1 of 2***

QSIG Category	Supported Features
Basic Call Setup	● Basic Call Setup ● Name and Number Transport ● Transit Counter
	1 of 2

Table 29: QSIG features supported by Avaya Communication Manager *2 of 2*

QSIG Category	Supported Features
Basic Supplementary Services	● Called/Calling/Busy/Connected Name and Number (Called/busy number is MSI only, see below) ● Name Identification Services ● Diversion (Call Forwarding) ● Diversion (Call Forwarding) with Reroute (using Path Replacement) ● Call Transfer ● Call Offer ● Call Completion (Automatic Callback) ● Centralized INTUITY AUDIX ● Path Replacement ● Call Transfer into QSIG Message Center
Value-Added (VALU) MSI (Also included with Basic Supplementary Services, but for Avaya systems only)	● Displays called party number to the calling party when the called number is ringing or busy (Called/Busy Number) ● Distinctive Ringing to identify internal/external and priority calls ● Call Coverage to networked switches. ● QSIG Leave Word Calling
Centralized Attendant	● Centralized Attendant Service (CAS)
	2 of 2

Note:

Although VALU-MSI only works with Avaya equipment, MSI information is passed through non-Avaya systems in an all-QSIG network. Thus, if you have two switches connected using QSIG through a non-Avaya switch, the MSI information still arrives at each end. Similarly, if two non-Avaya systems are sending their own MSI through an Avaya switch, and the connections are all QSIG, the Avaya switch sends on the information.

Basic call setup

Transit counter (TC)

Communication Manager provides QSIG TC as defined in ISO/IEC 6B032 and 6B033. It prevents infinite looping, connections giving poor transmission performance, and inefficient use of network resources.

TC is invoked automatically for ISDN basic calls and the Route Pattern screen indicates the number of switches through which a call may be routed.

Basic supplementary services

Called/calling/busy/connected name and number

Enables the calling party to see the name and number of the called party at the following times:

- While the call is ringing at the called party's terminal
- While listening to a busy tone because the called party's terminal was busy

Called/calling/busy/connected name is similar to the display provided for local on-switch calls, as well as for the DCS calls, with the following exceptions:

- Names longer than 15 characters are truncated; only the first 15 characters display.
- The number does not display unless QSIG VALU is enabled.

Name and number identification

Name and number identification allows a switch to send and receive the calling number, calling name, connected number, and connected name. Name and number identification displays up to 15 characters for the calling and connected name and up to 15 digits for the calling and connected number across ISDN interfaces.

You can administer outgoing calls as “yes”, “no”, or “restricted.” Restricted means that Communication Manager sends the information but sends it “presentation restricted,” which indicates to the receiving switch that the information should not be displayed. A received restricted number is included on the Call Detail Record (CDR).

Transit switch information

When Avaya equipment acts as a transit switch, the QSIG standards require it to pass on all supplementary service information that is not addressed to it. This includes name information. (A “transit” switch is a switch that routes an incoming call administered for Supplementary Services Protocol B to a trunk also administered for Supplementary Services Protocol B.) However, Basic Call Setup and number information is subject to modification by the transit switch. This means that trunk group administration on a transit switch does not override incoming name information, but may override incoming number information (as long as this does not lower the restriction on the information).

For example, if a non-restricted calling name and number are received by Avaya equipment acting as a transit switch, and if the outgoing trunk is administered for presentation restricted for both name and number, the number is passed on as “restricted” and name is passed on as “unrestricted.”

Tandem switch information

However, in the case of tandemed calls (calls involving two ISDN trunks that are not both administered for Supplementary Service Protocol B), trunk group administration may override both incoming name and number information, as long as doing so does not lower the restriction on the information.

For example, a tandemed call that comes in with restricted name information is sent out with restricted name information even if the outgoing trunk is administered for presentation unrestricted. However, non-restricted data is sent restricted if the trunk group administration is set for “presentation restricted.”

ISDN numbering formats

Numbering is specified on the ISDN Public-Unknown-Numbering and/or ISDN Private-Numbering screens. The numbering screen you use depends on how you administer the ISDN trunk group Numbering Format field.

However, if you format the Called Party Number with public numbering, the Calling/Connected Party Number is created in the public format even if you specify “private” on the ISDN trunk group screen. This provides the caller or called party a number that can be used to reach the other party. Since the call routes through the public network, the public Calling/Connected Party Number is a more accurate address.

Diversion (call forwarding)

Call forwarding works over a QSIG network.

When a call has already been forwarded 3 times over a QSIG trunk, it is not forwarded again but instead terminates at the final forwarded-to terminal. Remote activation and deactivation of this feature are not supported.

Diversion (call forwarding) with rerouting

A forwarded call can be rerouted in a private network to find a more cost-effective or resource-efficient path.

Note:

A forwarded call is typically not rerouted through the system that controls the forwarding party. This impacts certain features such as Call Coverage because that system no longer has control over the call. For example, the call cannot follow the forwarding user's coverage path if the forwarded call is not answered; instead, it follows the forwarded-to user's coverage path.

Call Transfer

QSIG Call Transfer is based on the current Communication Manager Transfer and Trunk-to-Trunk Transfer features. QSIG Transfer signaling occurs as long as one of the calls involves a QSIG trunk between the two switches.

Once a call is transferred, the transferring switch is unnecessary. Additional Network Feature-Path Replacement (PR) is invoked automatically to connect the transferred call more efficiently in the private network. QSIG Call Transfer attempts to connect the two parties more efficiently and drops the unnecessary switches.

QSIG Call Transfer provides the same functionality as the standard Transfer or Trunk-to-Trunk Transfer features, with additional call information available to the connected parties after the transfer completes.

Depending upon QSIG Identification Services administration, the connected parties' displays show each other's name and/or number. If the name and number are not available, the display of a connected party updates with the name of the involved trunk group.

Call Offer

This feature is the QSIG equivalent of Call Waiting.

A Private Telecommunication Network (PTN) offers up to four ways of invoking QSIG Call Offer (CO) (listed below). *Communication Manager uses only the first method*.

- Network invocation (immediate) — the PTN automatically invokes CO whenever the calling user makes a call to a user that is busy, if required by the service profile of the calling user.
- Consultation — the calling user, on being informed that a call has failed because it is busy at the destination and that CO may be possible, is able, within a defined time period (consultation timer), to request invocation of CO.

- Immediate invocation — the calling user is able to request invocation of CO as part of the initial call set-up.
- Network invocation (delayed) — the network, having informed the calling user that a call has failed because it is busy at the destination, invokes CO automatically unless the calling user initiates call clearing within a defined time period (automatic call offer invocation timer).

The effect of QSIG CO on the terminating end is similar to the DCS Call Waiting feature with the exception that for Call Waiting, the calling side (user or switch) does not have to convey any special message to invoke the feature. The Call Waiting Termination feature is driven based on the terminating user (for instance, single line analog set user with Call Waiting enabled).

For QSIG Call Offer, the system takes advantage of the additional information available from the far end, if QSIG Call Offer invokes successfully, and provides similar information to the calling user as the Call Waiting feature provides for internal calls, with the exception that the display update will be “offered” instead of “wait” to reflect invocation of QSIG Call Offer service.

On successful invocation of the QSIG Call Offer service, the system provides the following:

- To the busy analog set user, the same tone as Call Waiting Termination feature.
- To the busy multi call-appearance set (for instance, at least one call-appearance is busy for an active call and at least one call-appearance is available for incoming calls) user, the available appearance rings normally.

For incoming QSIG calls, the QSIG Call Offer service may use path retention which is a generic mechanism to retain the signaling connection so that the originating party can decide whether to invoke the supplementary service. The network connection can be retained for more than one of the supplementary services for which path retention has been invoked.

Call Completion

Completion of Calls to Busy Subscribers (CCBS) and Completion of Calls on No Reply (CCNR) are the equivalent QSIG features of Automatic Callback (ACB) on busy and ACB on no answer, respectively.

An analog voice terminal user activates CCBS or CCNR by pressing the **Recall** button or flashing the switchhook and then dialing the Automatic Callback (ACB) activation feature access code. An analog user can activate only one ACB call at any given time.

A multi-appearance voice terminal user can activate CCBS or CCNR for the number of ACB buttons assigned to the terminal.

CC options

QSIG CC has the following major options that are negotiated between the Originating and the Terminating switch:

- Path reservation — there are two methods of establishing the CC call:
  - Path reservation method
  - Non-reservation method
- Retention of signaling connection — there are two ways in which CC uses call independent signaling connections:
  - Connection retention method
  - Connection release method
- Service retention — there are two possible behaviors when User B is found to be busy again after User A responds to CC recall:
  - Service retention method
  - Service cancellation method

As an originating switch for QSIG CC, Communication Manager selects the following major options:

- Non-reservation method for the Path Retention option
- Connection release method for the retention of signaling connection option
- Service cancellation method for the Service Retention option

As a Terminating switch for QSIG CC, Communication Manager selects the following major options:

- Non-reservation method for the Path Retention option
- Either the connection release method or the connection retention method for the retention of signaling connection option, depending on which the originating switch requests.
- Service cancellation method for the Service Retention option

Path Retention

Path Retention is a generic mechanism for retaining a network connection that can be used by supplementary services during call establishment.

The originating switch invokes path retention for one supplementary service or for several simultaneous supplementary services. Invoking a particular supplementary service means retaining the network connection, if the terminating switch encounters the appropriate conditions. The originating switch is informed of the reason for retaining the connection. It then decides (for example, by consulting the calling user) whether to invoke the supplementary service. Under some circumstances, in which the network connection is retained, more than one of the supplementary services for which path retention has been invoked may be applicable.

Successive retentions of the network connection by the terminating switch following a single path-retention invocation by the originating switch are possible. This is a result of different conditions being encountered at the terminating switch. When an attempt is made to invoke a supplementary service for which the network connection has been retained, a further condition can be encountered that can cause the network connection to be retained again for the same or a different supplementary service.

Path retention is specified in terms of a Path Retention entity existing within the coordination function at the originating switch and at the terminating switch.

QSIG Centralized INTUITY AUDIX

QSIG allows users on a remote node (served user switch) to “cover” to an INTUITY AUDIX system on another node (message center switch). The original calling party information, called party information, and reason for coverage is provided to the INTUITY AUDIX system so that each is identified properly during message recording/retrieval.

To use a centralized INTUITY AUDIX system, you must use QSIG Diversion. On a served user switch, the call goes to call coverage using Diversion to the hunt group assigned to the INTUITY AUDIX system on the message center switch. Then the message center switch sends all the appropriate information to the INTUITY AUDIX system so that it correctly answers the call.

QSIG Centralized INTUITY AUDIX also uses path optimization using QSIG Diversion with Reroute.

Transfer into INTUITY AUDIX works when transferring from a served-user switch into an INTUITY AUDIX system at the message center switch.

What you get with QSIG Centralized INTUITY AUDIX

- Calls to users on a branch cover or forward correctly and are answered by the INTUITY AUDIX system:
  - With a personalized greeting
  - With an appropriate busy or not available greeting, depending upon the reason the call was redirected

  Caller can leave a message for the called party.
- Once a subscriber logs into the INTUITY AUDIX system (by dialing the INTUITY AUDIX number and entering the extension and password), the subscriber can perform the following activities:
  - Listen to or delete messages (voice, fax, or text).
  - Leave a message for other subscribers on the same INTUITY AUDIX system without calling them.
  - Forward a message to another subscriber on the same INTUITY AUDIX system.

  - Access the INTUITY AUDIX directory to address a message (*A).
  - Access the INTUITY AUDIX directory to find a subscriber's extension (**N).
  - Record or change his/her greeting.
  - Transfer out of INTUITY AUDIX system (*T or 0).

- Message Waiting Indication (typically a lamp, but may also be a stutter dial tone or display) indicates the presence of new messages.

  If another vendor's system, acting as a served user switch, does not provide this functionality, the end user will not receive an MWI indication.I

- When a remote subscriber logs in to an INTUITY AUDIX system from the subscriber's phone, the subscriber does not need to enter his or her extension.

  Instead of entering the extension, *, the password, and *, the subscriber can enter *, the password, and then #.

- Leave Word Calling works for users on a single switch, and across served user switches.

  With Release 11 or newer software, LWC will work across a QSIG network.

What you do not get with QSIG Centralized AUDIX

- With Release 11 or newer software, Transfer into INTUITY AUDIX does not work from a served-user switch.

Other QSIG Centralized Messaging

With a QSIG centralized messaging system, the remote switch is called a served user switch. The messaging system connected to the network using the QSIG protocol is called the message center switch. The Octel Serenade is such a messaging system.

QSIG allows an Avaya switch to be a served user switch of a non-Avaya message center switch. Therefore, when the messaging system is the message center switch, it can serve the Avaya switch if that messaging system has a QSIG interface. The Octel Serenade is such a messaging system.

For users in a QSIG messaging network, only one message center can be administered for each Avaya served user switch on all-Avaya platforms.

With path optimization using QSIG Diversion with Reroute, the system will attempt to reroute a call when the following options are enabled:

- ISDN-PRI or ISDN-BRI (qsig-mwi type of hunt group)
- QSIG Basic Call Setup
- QSIG Supplementary Services with Rerouting

Path Replacement

Path Replacement (PR) is the process of routing an established call over a new, more efficient path, after which the old call is torn down leaving those resources free. Path Replacement offers customers potential savings by routing calls more efficiently, saving resources and trunk usage.

Path Replacement occurs with Call Transfer, and in the following cases:

- Call forwarding by forward switching supplementary service, including the case where Call Diversion by Rerouting fails, and call forwarding is accomplished using forward switching
- Gateway scenarios where Avaya equipment, serving as an incoming or outgoing gateway, invokes PR to optimize the path between the gateways
- Calls in queue/vector processing even though no true user is on the call yet
- QSIG Look-ahead Interflow call, Best Service Route call, or adjunct route

⚠ **CAUTION:**

Depending on the version of Call Management System (CMS) you are using, some calls may go unrecorded if you administer your system for Path Replacement in queue/vector processing. Please see your Avaya representative for more information.

Communication Manager provides QSIG Path Replacement (PR) as defined in ISO/IEC 13863 and 13874. With this feature, a call's connections between switches in a private network can be replaced with new connections while the call is active.

PR is invoked when a call is transferred and improvements may be made in the routes. For example, after a call is transferred, the two parties on the transferred call can be connected directly and the unnecessary trunks are dropped off the call.

PR requires Rerouting (RR) to be turned on in both switches. The routing administered at the endpoints allows for a more efficient route connection. In some cases, where all or some of the original route is the most effective route, Path Retention is invoked.

PR selects the best route based on the preference assigned to routes in the route pattern form. Least cost Supplementary Service B (SSB) routes must be first, followed by more expensive routes.

Note:

When routes to SSB trunks are included with routes to non-SSB trunks, SSB trunks must appear first on the Route Pattern screen. This is because as soon as PR encounters a non-SSB trunk in the route pattern, it stops looking.

Class of Restriction (COR) and Facility Restriction Levels (FRL) are followed in routing calls. PR is not invoked on data calls because there is a period of time when information can be lost.

Transfer into QSIG Message Center

This feature uses QSIG Call Transfer, along with a manufacturer-specific information (MSI) message, to transfer a call directly into a subscriber’s mailbox, when the voice mail system is connected to the served user switch by a QSIG link. The voice mail system must be an Avaya system that supports the QSIG transfer into QSIG Message Center MSI operation.

Note:

This feature currently works only with the Octel 200/300 Serenade voice mail system. This feature does not work with a QSIG Centralized INTUITY AUDIX system, unless the system is at R11 or newer.

The entire route must be QSIG, from the switch activating Transfer into Message Center to the message center switch/ voice mail system.

Value-Added (VALU) MSI

Value-Added (VALU) Manufacturer-specific information (MSI) adds the following feature transparency to QSIG networks:

- Called/busy number — The system sends and displays across the network the called party’s number to the calling party during alerting. It updates the display to “connected number” when the called party answers the call. It also sends and displays a busy party’s number. This serves to confirm to the caller that he or she dialed the correct number.

  The called/busy number feature is an extension to QSIG called/busy name. For additional information, see Name and number identification on page 189.

  The called/busy number never displays alone; it displays only if the called/busy name is available (for instance, received from the far end and marked “presentation allowed”). In contrast, upon receipt of a calling number without a calling name, the number displays with the words “CALL FROM.”

- Distinctive ringing — QSIG VALU provides two kinds of distinctive ringing across the network: internal and external.
- Call coverage — The system allows calls to be covered by extensions across the network. This coverage operates similarly to DCS Call Coverage, though the connectivity of the network itself differs. If administered, path replacement is invoked after coverage.

QSIG Centralized Attendant Services (CAS)

The CAS feature enables one or more branches to concentrate their attendants on one main. CAS provides most features that are normally available to the basic attendant service between switches. All current QSIG features are available with CAS. CAS functionality is enabled through the **Centralized Attendant** field on the **Customer Options** form (page 7, QSIG Optional Features).

Note:

If the **Centralized Attendant** field is **y**, the **IAS Branch** field on the **Console Parameters** form does not appear.

Note:

QISG-CAS does not interwork with RLT-CAS.

Potential CAS limitations

There are a few potential limitations when using CAS:

- Path Replacement does not work immediately.

  This means that resources are being utilized longer with CAS.
- Path Replacement is not guaranteed.
- Path Replacement does not enable a branch to act as a gateway.
- No Path Replacement functionality takes place during or after a conference.

What are the CAS functions

The following are CAS functions:

- Attendant-seeking calls at a branch reach the attendant at the main.
- Attendant splitting away and calling the extended to party.
- Night service.
- Monitoring agents, per attendant group.
- Announcements for attendant seeking calls.
- Attendant calls enter the attendant queue, with priorities (calls that originate from the branch do not have different priorities in the queue).
- Attendant display of user’s COR.
- Attendant split/swap.
- Path Replacement after a transfer.
- Attendant return call (release loop operation, returns to same attendant, if available; if not, then the attendant group).

- Display enhancements.
- Attendant conference.

Call-independent Signaling Connection (CISCs)

A Call-independent Signaling Connection (CISC**)** provides a temporary signaling path through ISDN switches for exchanging supplementary service information; for example, exchange Facility Information Elements in call control messages, FACILITY messages, or a combination of both on ISDN D-channels. There is no B-channel related to the connection; no data or voice transmissions take place.

CISCs are administered in the same way as Non-Call Associated Temporary Signaling Connections (NCA-TSCs).

About Non-Call Associated Temporary Signaling Connection (NCA-TSC)

A Non-Call Associated Temporary Signaling Connection (NCA-TSC) is a connection not related with any **ISDN** B-channel connections. Communication Manager supports two types of NCA-TSC that conform to two different protocol standards:

- The *non-QSIG* type of NCA-TSC is used for the DCS over ISDN-PRI D-channel and DCS AUDIX applications. Only ISDN-PRI signaling groups administered with supplementary service protocol **a** support AT&T and WorldCom NCA-TSCs.
- The *QSIG* type of NCA-TSC is required for certain QSIG features such as Call Completion (Automatic Call Back). This type of NCA-TSC is referred to in the QSIG protocol standards as a Call-Independent Signaling Connection (CISC). Only ISDN-PRI signaling groups administered with supplementary service protocol **b** support QSIG NCA-TSCs. In addition, BRI trunk D-channels support QSIG NCA-TSCs.

  Note:

  > You will not see a second page (Administered NCA-TSC Assignment) on the Signaling Group screen when you set the supplementary service protocol to **b** for QSIG.

An NCA-TSC for QSIG is not administered ahead of time, but is invoked dynamically by the QSIG feature that needs it. Some QSIG features remove the NCA-TSC when it is no longer needed; others leave it active for a longer period of time.

Administering QSIG

The QSIG administration procedures include:

- Basic QSIG administration
- Administering QSIG supplementary services
- Administering Centralized Attendant Services
- Administering QSIG VALU Call Coverage

Basic QSIG administration

To set up basic QSIG:

1. Verify with your Avaya sales representative or project manager what QSIG capabilities the system should have. The capabilities in Table 30 apply:

Table 30: QSIG capabilities ***1 of 2***

Capability categories	Cross-networking features
QSIG basic	● Calling/connected name and number ● Calling name and number identification ● Transit Counter
	1 of 2

Table 30: QSIG capabilities ***2 of 2***

Capability categories	**Cross-networking features**
Basic supplementary service	● Called/busy name ● Called/calling name/number delivered to and received from DCS networked switches ● Call Completion ● Call Forwarding (Diversion) ● Calling Name Identification ● Call Offer ● Centralized INTUITY AUDIX ● Call Transfer ● Path Retention ● Message Waiting Indication ● Diversion (call forwarding) with rerouting ● Path Replacement ● Transfer into QSIG Voice Mail ● QSIG/DCS+ Voice Mail Interworking
Value-Added (VALU) MSI (Also included with basic supplementary services, but for Avaya equipment only)	● Called/busy number display ● Distinctive ringing ● Call Coverage ● Leave Word Calling
Centralized Attendant Service	● Centralized Attendant
	2 of 2

Note:

Although VALU-MSI only works with Avaya equipment, MSI information is passed through non-Avaya systems in an all-QSIG network. Thus, if you have two switches connected using QSIG through a non-Avaya switch, the MSI information still arrives at each end. Similarly, if two non-Avaya systems are sending their own MSI through an Avaya switch, and the connections are all QSIG, the Avaya switch sends on the information.

2. Determine whether the system is using ISDN-PRI, ISDN-BRI, or ATM for the QSIG network connections. Your sales representative or project manager should know this. (If the system is using ATM trunking for QSIG, see *ATM Installation, Upgrades, and Administration* (555-233-124).
3. Enter **`display system-parameters customer-options`** on the SAT command line of your system administration screen.
4. On page 1, verify fields as follows:
   - **G3 Version** field is **V11** or later.
5. If the system is using ATM for QSIG, go to page 3 and verify the following field:
   - **Async. Transfer Mode (ATM) Trunking** field is *y*.
6. On page 4, verify fields as follows:

   If the system is using ISDN-BRI for QSIG:
   - **ISDN-BRI Trunks** field is *y*.

   If the system is using ISDN-PRI for QSIG:
   - **ISDN-PRI** field is *y*.

   If the system is using QSIG supplementary services with or without rerouting:
   - **Restrict Call Forward Off Net** field is *n* (page 5).
7. On page 8, verify fields as follows:
   - **Basic Call Setup** field is *y*.

   If the system is using QSIG supplementary services:
   - **Basic Supplementary Services** field is *y*.

   If the system is using QSIG supplementary services with rerouting:
   - **Supplementary Services with Rerouting** field is *y*.

   If the system is using QSIG VALU:
   - **Value-Added (VALU)** field is *y*.
8. (For ISDN-PRI only). Administer or check the QSIG DS-1 circuit pack. Check for the following field entries:

   When connecting two Avaya switches:
   - **Connect** field - *`pbx`*.
   - **Interface** - *`user`* or *`network`*.
   - **Country protocol** - *`1`*.
   - **Protocol version** - *`a`*.
   - **Signaling mode** - *`isdn-pri`* or *`isdn-ext`*.

- **Channel numbering (E1)** - *sequential* or *timeslot* (This item must match between the local switch and the receiving switch. If NFAS is used, this must be *timeslot*).

When connecting an Avaya switch to another vendor's product:

- **Connect** field - *pbx*.
- **Interface** - *peer-master* or *peer-slave*.
- **Peer protocol** - *q-sig*.
- **Signaling mode** - *isdn-pri* or *isdn-ext*.
- **Channel numbering (E1)** - *sequential* or *timeslot* (This item must match between the local switch and the receiving switch. If NFAS is used, this must be *timeslot*).

9. (For ISDN-BRI only). Administer or check the QSIG ISDN-BRI circuit pack.
10. Administer or check the QSIG ISDN trunk group(s) (PRI or BRI) connected to the DS-1 or BRI circuit pack. Check for the following field entries:

On page 1:

- **Group Type** - *isdn*
- **Supplementary Service Protocol**- *b* or *d* where:

b	ISO QSIG standards (including the ETSI Version 2 and European Computer Manufacturer's Association (ECMA) standards aligned with the ISO standards)
d	ETSI Version 1 and ECMA standards issued prior to the ISO standards for QSIG private network (supports only Name Identification and Additional Network Feature Transit Counter (TC))

- **Outgoing Display** - *y*
- **QSIG Value-Added** - *y*

On page 2:

- **Hop Dgt** - *y*
- **Disconnect Supervision** - *y*
- **Numbering Format** - select from *public*, *private*, *unknown*, *unk-pvt*
- **NCA - TSC Trunk Member** - The trunk member whose D-channel routes CISCs.
- **Send Called/Busy/Connected Number** - *y*
- **Send Calling Number** - *y*
- **Send Name** - *y*
- **Path Replacement with Retention** - *y*

Administering QSIG supplementary services

To set up QSIG supplementary services:

1. Administer or check the ISDN **Numbering - Public/Unknown** screen, as necessary.
2. Administer or check the ISDN **Numbering - Private** screen, as necessary.
3. Administer or check the **Signaling Group** screen, as necessary.

   Check for the following field entries to ensure proper operation of Call Completion:

   - **Supplementary Service Protocol** - **b**
   - **Max Number of NCA TSC** - greater than **0**
4. Administer or check the **Route Pattern** screen, as necessary.

   Check for the following field entries to ensure proper operation of Call Completion and Transit Counter:

   - **TSC** - **y** (necessary if switch is a transit node for TSC)
   - **Hop Lmt** - between **1** and **32**
5. Administer or check the **Feature-Related System Parameters** screen.

   Check for the following field entries to ensure proper operation of Call Completion and Call Transfer:

   - **Trunk-to-Trunk Transfer** - *all* (page 1)
   - **QSIG TSC Extension** - valid extension number to serve as TSC for both incoming and outgoing QSIG network calls (page 8).
   - **Automatic Callback - No Answer Timeout Interval** (rings) - enter the number of times, **2** to **9**, a callback call should ring at the caller’s phone before the callback is cancelled (page 1).
   - (For AUDIX only) MWI - **Number of Digits per AUDIX Subscriber** (page 8) - enter the number of digits in messaging subscriber extensions, if any.

     The value in this field must match the value of the **Extension Length** field on the **Switch Interface Administration** screen of the AUDIX system.
   - (For Octel Serenade and Aria) **Number of Digits Per Subscriber** is set by the leading digit.

     Please refer to your Octel documentation for more information.
   - (For AUDIX/Octel Serenade support only) **Unknown Numbers Internal for AUDIX** - *y* if, when the switch cannot identify a calling number as internal or external, the switch should treat it as internal for AUDIX use (page 8).

6. Administer or check the **Class of Service (COS)** screen for each COS that may be using the QSIG network.

   Check for the following field entries to ensure proper operation of Auto Callback, Call Offer, and Call Forward:

   - **Restrict Call Forward Off-Net** - **n**
   - **Auto Callback** - **y**
   - **QSIG Call Offer Originations** - **y**

Call completion administration

In addition to the Basic QSIG Supplementary Services administration described above, complete the following administration.

To administer for call completion:

1. On the **Trunk Group** screen, page 1, set the **Supplementary Service Protocol** field to *b* and administer the trunk for Call Independent Signalling Connections (NCA-TSCs).

Transfer into Avaya QSIG Message Center (Octel Serenade only)

In addition to the Basic QSIG Supplementary Services administration described above, complete the following administration

To transfer into Avaya ASIG message center (Octel Serenade only):

1. On the **System-Parameters Customer-Options** screen, page 8, the **Transfer Into QSIG Voice Mail** field must be set to *y*.
2. On the **Feature Access Code (FAC)** screen, page 4, assign a Feature Access Code in the **Transfer to Voice Mail Access Code** field.
3. A hunt group must be in the coverage path of the user’s mailbox to be transferred into, as administered on the **Station and Coverage Path** screens.

   On the **Hunt Group** screen, page 2, for this hunt group, **qsig-mwi** must be entered in the **Message Center** field and the number for the voice mail system must be entered in the **Voice Mail Number** field.

QSIG/DCS+ Voice Mail Interworking

QSIG/DCS Voice Mail Interworking requires Release 9 or later software. Also, on page 8 of the **System-Parameters Customer-Options** screen, the **Interworking with DCS** field under **QSIG Optional Features** must be enabled. This feature allows an INTUITY AUDIX system, or an Octel Serenade system to act as a centralized voice mail server in a DCS/QSIG mixed network environment.

Administering Centralized Attendant Services

Note:

An attendant console must be administered at the main, before administering Centralized Attendant Services. See the *Administrator's Guide for Avaya Communication Manager (555-233-506)* for instructions on administering an attendant console.

To administer centralized attendant services:

1. Enable QSIG **Supplementary Services with Rerouting** on the **System-Parameters Customer-Options** screen, page 8, as described above.
2. On the **System-Parameters Customer-Options** screen, page 8, enter **y** in the **Centralized Attendant** field.
3. On the **Console Parameters** screen, enter **QSIG-main** or **QSIG-branch** in the **CAS** field.
   a. If **QSIG-branch** is entered in the **CAS** field, then enter a number in the **QSIG CAS Number** field.
   b. If **QSIG-branch** is entered in the **CAS** field, then the **AAR/ARS Access Code** field is optional.
4. On the **QSIG ISDN Trunk Groups** screen, enter **b** for the **Supplementary Service Protocol** field.
5. Assign an extension to **attd** on the **Dial Plan Analysis** screen at the main switch.
6. Administer each QSIG Supplementary Service that will be used by attendants.

Administering QSIG VALU Call Coverage

To set up QSIG VALU Call Coverage:

1. Enable (**y**) the **QSIG Basic Supplementary Services** field on the **System-Parameters Customer-Options** screen, page 8, described above.
2. Enable (**y**) **Value-Added (VALU)** on the **System-Parameters Customer-Options** screen, page 8, as described above.
3. On a **Trunk Group** screen, page 1 enter **y** in the **QSIG Value-Added** field, and enter **b** in the **Supplementary Service Protocol:** field.

4. Administer the **System-Parameters Call-Coverage/Call-Forwarding** screen as normal, with the inclusion of the following fields:

   - **Immediate Redirection on Receipt of PROGRESS Inband Information**, page 1 — Enter **y** to speed up redirection of subsequent coverage points or call processing.

     This may be necessary in cases where coverage path endpoints over non-Avaya switches are unavailable but the QSIG networked switch (or the public network) sends PROGRESS messages that delay the local switch from redirecting the call elsewhere. If the QSIG network contains only Avaya switches, enter n.

   - **QSIG VALU Coverage Overrides QSIG Diversion with Rerouting**, page 1 — Enter **y** to ensure that the "coverage after forwarding" activation/deactivation defined at a user's phone (on the **Station** screen) takes precedence over the system-wide "coverage after forwarding" activation/deactivation selection (on the **System Parameters Call Coverage/Call Forwarding** screen).

     With QSIG Diversion with Rerouting active, the system-wide selection takes precedence unless you enter **y**.

     Table 31: QSIG Diversion with Rerouting examples on page 207 lists examples of these field values and a description of the rerouting pattern.

Table 31: QSIG Diversion with Rerouting examples ***1 of 2***

Field			
Cvg. After Fwd (Station screen)	**Cvg. After Fwd (System Parameters - Coverage screen)**	**QSIG VALU Coverage Overrides QSIG Diversion**	**Rerouting pattern**
y	n	n	Call does not go to local user's coverage after failed forward attempt. Call control passed to switch to which call forwarded.
y	n	y	Call goes to local user's coverage after failed forward attempt.
			1 of 2

Table 31: QSIG Diversion with Rerouting examples ***2 of 2***

Field			
Cvg. After Fwd (Station screen)	**Cvg. After Fwd (System Parameters - Coverage screen)**	**QSIG VALU Coverage Overrides QSIG Diversion**	**Rerouting pattern**
n	y	n	Call goes to local user's coverage after failed forward attempt.
n	y	y	Call doesn't go to local user's coverage after failed forward attempt. Call control passed to switch to which call forwarded.
			2 of 2

Note:

If the **Maintain SBA at Principal** field is enabled (**y**), then Path Replacement is disabled.

5. Define the remote QSIG users that you may include in coverage paths using the **Remote Call Coverage Table**.

   See "Defining Coverage for Calls Redirected to External Numbers" in the "Handling Incoming Calls" chapter of the *Administrator's Guide for Avaya Communication Manager (555-233-506)*. See also the **Remote Call Coverage Table** screen in the same book.

6. Define coverage paths for users as required.

QSIG-related phone administration

As you set up each user's phone, QSIG networking features allow the following.

- QSIG displays the user's name as entered in the **Name** field on the **Station** screen, both on the display of another networked phone when called by that user or when calling that user.
- QSIG allows call waiting from networked phone calls if you set the **Call Waiting Indication** field to **y**.
- QSIG allows auto callback from networked phones if you create an auto callback button for the user.

QSIG-related Hunt Group administration

As you set up each hunt group, you must enter either **grp-name** or **mbr-name** in the **ISDN Caller Disp** field, page 1. This entry determines which of the following the system displays on a QSIG networked phone that calls the hunt group:

- The hunt group name/extension
- The hunt group member’s name/extension

QSIG-related Terminating Extension Groups administration

As you set up each terminating extension group, you must enter either **grp-name** or **mbr-name** in the **ISDN Caller Disp** field. This entry determines which of the following the system displays on a QSIG networked phone that calls the terminating extension group:

- The group name/extension
- The group member’s name/extension

QSIG-related AUDIX/Message Center administration

Follow these steps to set up Related Administration of AUDIX/Message Centers.

Note:

Set up QSIG TSCs before you administer messaging. See Call Completion.

To administer QSIG-related AUDIX/Message Center:

1. (Local node message center switch only) Complete the **Processor Channel Assignment** screen.
2. (Local node message center switch only) Complete the **Message Waiting Indication Subscriber Number Prefixes** screen.
3. (Local node message center switch only. This requires bx.25 or C-LAN integration) Complete the Station screen as specified in the AUDIX documentation.

   Verify the following field entry:

   - **MWI Served User Type** - **qsig-mwi**

4. (Served user switch only) On the **Hunt Group** screen, set the following fields for the AUDIX hunt group:

    - **Message Center** - **qsig-mwi** (page 2)
    - Voice Mail Number and Routing Digits (for example, AAR/ARS Access Code):

      Digits entered in these fields should be selected so that the processing of these digits by the served user switch results in a call being redirected to the message center switch by an ISDN-PRI supplementary service protocol **b** facility. For example, if the message center switch is an Avaya switch, the digits entered should reroute the call to the AUDIX hunt group on the message center switch.
    - **Calling Party Number to AUDIX** - **y**

Migrating to QSIG: some considerations

If you are planning to migrate your network from DCS to QSIG, then there are some issues you need to consider. The following is a list of some of the issues:

- Feature Parity
- Virtual Private Networking
- Voice Messaging Integration
- DCS/DCS+ and QSIG Interworking

This section offers only an overview of the above issues. For more details, please contact your Avaya representative.

Feature Parity

The QSIG protocol was created as a set of standards and specifications for interoperability in multi-vendor network environments. This was a response to the many proprietary protocols (such as Avaya’s DCS) which did not interoperate among vendors.

In order to ensure that features exclusive to proprietary protocols would not be lost when a network is migrated to QSIG, vendors are able to create Manufacturer Specific Information (MSI) messages. By QSIG standards, these messages are passed on, unchanged by any intermediate switches in a network, even if the intermediate switches are from a different manufacturer than the sending and terminating ones.

Communication Manager has MSI features that emulate DCS features not standard with QSIG.

Virtual Private Networking

Some telecommunication companies have provided for DCS+ Virtual Private Networking by transporting Temporary Signaling Connection (TSC) messages in their public switched ISDN networks. There are currently no such provisions by any service provider for QSIG Call Independent Signaling Connections.

Voice Messaging Integration

Before migrating from DCS to QSIG, it is important to know whether or not the existing messaging infrastructure will support integration with QSIG networking.

Mode Code signaling is a common integration method, but is not supported over QSIG networking in Communication Manager. Mode Code signaling uses ISDN tandem trunk signaling to pass messages, but this ISDN signaling was not made to interwork with QSIG.

DCS/DCS+ and QSIG Interworking

Migration of segments of the network, as opposed to all at once, is feasible. However, there is limited interworking functionality between DCS+ and QSIG, and no interworking functionality between traditional DCS and QSIG.

The following features may be interworked between a DCS+ network and QSIG:

- Basic call with name and number
- Leave word calling (LWC)
- Message waiting indication (MWI)
- Centralized voice mail

Note:

For DCS+ leave word calling interworking with QSIG, all systems must be running Communication Manager. LWC activates MWI lamps on Avaya phones only.

Centralized Attendant Service

The main Centralized Attendant Service (CAS) topics in this section are

- What is Centralized Attendant Service (CAS)
- Administering CAS

Note:

See Chapter 5: Feature interactions and considerations for feature interaction information and other considerations when using CAS.

What is Centralized Attendant Service (CAS)

Centralized Attendant Service allows attendants in a private network of switching systems to be concentrated at a central or main location. Thus, CAS reduces the number of attendants required at a branch. For example, a chain of department stores can have a centralized attendant location at the main store to handle calls for the branch stores.

Each branch in a CAS has its own Listed Directory Number (LDN) or other type of access from the public network. Incoming trunk calls to the branch, as well as attendant-seeking voice terminal calls, route to the centralized attendants over release link trunks (RLT).

The CAS attendants are at the main location. The CAS main switch operates independently of the CAS branch switches. Operation for CAS main-switch traffic is identical to operation of a stand-alone switch.

A branch in a CAS network can connect to only one main. Each branch connects to the main by way of RLTs. These trunks provide paths for:

- Sending incoming attendant-seeking trunk calls at the branch to the main for processing and extending them 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
- Routing calls from the branch to the main

This following sub-sections cover the topics:

- CAS Queues
- CAS Backup Service
- CAS Remote Hold
- Branch-generated call-identification tones
- CAS Outgoing Call Routing
- CAS Incoming Call Routing

CAS Queues

Two queues are associated with CAS calls: one at the main and one at the branch. If idle RLTs are available from the branch to the main, RLTs are seized and CAS calls are queued at the main along with other attendant-seeking calls. If all RLTs are in use, CAS calls to the attendant are queued at the branch in a RLT queue. The length of the queue can vary from 1 to 100, as set during administration of the RLT group.

CAS Backup Service

Backup service sends all CAS calls to a backup extension in the branch if all RLTs are maintenance-busy or out of service, or if the attendant presses a backup button that is not lighted.

To activate the CAS backup service feature and provide notification that backup service is in effect:

1. Assign the backup extension to a Backup button and associated status lamp.

   The status lamp remains lighted as long as backup service is in effect.

To deactivate the CAS backup service feature:

1. The attendant presses the Backup button while the status lamp is lighted.

Calls are not sent to the backup extension unless all RLTs are maintenance-busy or out of service.

CAS Remote Hold

The attendant can put a CAS call from a branch on Remote Hold. The branch holds the call and drops the RLT. After a time-out (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 use Remote Hold when they have to put a call on hold to keep RLTs from being tied up unnecessarily.

Branch-generated call-identification tones

The branch in a CAS network generates call-identification tones and transmits them to the CAS attendant by way of the RLT. These tones indicate the type of call coming 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 being connected to the caller. The tones may vary by country. See *Console Operations* for information on these tones.

CAS Outgoing Call Routing

The centralized attendant at the main has access, through RLTs, to all outgoing trunk facilities at the branches in a CAS network. The attendant can extend an incoming LDN call to an outgoing trunk at a branch by dialing the access code and allowing the caller to dial the rest of the number or by dialing the complete outgoing number.

CAS Incoming Call Routing

Calls extended to busy single-line voice terminals at the branch wait automatically. If there is a call in queue, the user hears a busy signal. When station hunting and send all calls is administered, the call routes along the administered path. Not answering any waiting extended call within an administered interval causes the branch switch to return the call to the attendant. Call Waiting does not apply to multiappearance terminals; if no appearances are available, busy tone is sent to the attendant, who tells the caller that the line is busy.

Calls from voice terminals at the branch to an attendant also route over RLTs seized by the branch switch. A branch caller reaches the attendant by dialing the attendant-group access code. The 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.

If an extended call returns to the main attendant unanswered, the called party at the branch does not drop but continues to be alerted until the caller releases. This allows the attendant to talk to the caller, then extend the call again, if the caller wishes, without redialing the number.

Administering CAS

Table 32: CAS administration on page 215 lists the screens and fields values to administer CAS.

Table 32: CAS administration

Screen	Field
Attendant Console	Page 3: ● Feature Button Assignments — cas-backup -trunk-name
Console-Parameters	Page 1: ● CAS ● RLT Trunk Group Number ● CAS Back-Up Ext Page 2: ● Timed Reminder on Hold ● Return Call Timeout (sec)
Station (multi-appearance)	● Feature Button Assignments — cas-backup — flash — trunk name — night serv
Trunk Group (RLT)	● All
Feature Access Code (FAC)	Page 1: ● CAS Remote Hold Access Code

Extended Trunk Access

The main Extended Trunk Access (ETA) topics in this section are

- What is Extended Trunk Access (ETA)
- Administering Extended Trunk Access
- About Extended Trunk Access interactions

What is Extended Trunk Access (ETA)

Use Extended Trunk Access in conjunction with Uniform Dial Plan (UDP) to allow a switch to send any unrecognized number (such as an extension not administered locally) to another switch for analysis and routing. Such unrecognized numbers can be Facility Access Codes, Trunk Access Codes, or extensions that are not in the UDP table. Non-UDP numbers are administered on either the First Digit Table (on the **Dial Plan Record** screen) or the Second Digit Table. They also are not administered on the ETA Call Screening Table. ETA helps you make full use of automatic routing and UDP.

Historically, ETA has been used by satellite switches to access stations, trunks, and features at the main switch. ETA frees you from having to enumerate the entire dial plan for the main or satellite complex. Calls that would get intercept treatment without ETA are routed to a remote switch to be reprocessed. The following processing takes place when ETA is administered:

- ETA call is identified because it fails all other routing possibilities.
- The dialed string is not in the ETA Call Screening Table.
- An available route pattern is selected based on the **Dial Plan** screen **ETA Routing Pattern** or **ETA Node Number** entries.
- The dialed string is sent to the remote switch.

Administering Extended Trunk Access

Screen	Field
Dial Plan Parameters	● ETA Routing Pattern ● ETA Node Number
ETA Call Screening Table	● Call Screening Entry

⚠ **CAUTION:**

Switches can be chained together using ETA. However, you must ensure that switches do not route in a circular ETA call setup. Switch A can route to switch B, and switch B can route to switch C. But, if switch A routes to switch B and switch B routes to switch A, you create a circular ETA call setup.

Examples of ETA administration

CASE #1

- **ETA Route Pattern** — Not administered
- **ETA Node Number** — Not administered

In this case, **ETA** is not active. It is not used to route undefined dialed strings.

CASE #2

- **ETA Route Pattern** — Administered
- **ETA Node Number** — Not administered

In this case, the **ETA Route Pattern** routes undefined dialed strings. However, since an **ETA Node Number** is not specified, non-call-related DCS messages are not routed.

CASE #3

- **ETA Route Pattern** — Not administered
- **ETA Node Number** — Administered

In this case, the **ETA Node Number** provides the route pattern. Non-call-related DCS messages also can route since a node number is supplied.

CASE #4

- **ETA Route Pattern** — Administered
- **ETA Node Number** — Administered

In this case, the **ETA Route Pattern** routes undefined dialed strings while the **ETA Node Number** routes DCS messages. Nodes themselves do not have to be administered for ETA. ETA should not be used over tandem-tie trunks.

About Extended Trunk Access interactions

- Abbreviated Dialing

  Abbreviated Dialing calls are routed by means of ETA.
- Attendant

  Attendants calls are routed by means of ETA.
- Data-Call Setup

  Analog and digital endpoints can access ETA. The digit string goes to the remote switch like any other digit string is sent. The remote switch handles the data-call setup from that point forward.
- Facility Restriction Levels

  It is possible to restrict trunks that are being used in conjunction with ETA by assigning FRLs.
- Last Number Dialed

  If a number is routed by means of ETA to a remote switch and you want to reaccess that number, then reaccess uses ETA.
- Modem Pooling

  Modems in Modem Pools are treated like all other trunks.
- Remote Access

  Remote-access trunks are able to access the ETA feature just as any other trunk or station does.

Inter-PBX Attendant Service

The main Inter-PBX Attendant Service (IAS) topics included in this section are

- What is Inter-PBX Attendant Service (IAS)
- Administering Inter-PBX Attendant Service
- About Inter-PBX Attendant Service interactions

What is Inter-PBX Attendant Service (IAS)

Inter-PBX Attendant Service allows attendants for multiple branches to be concentrated at a main location. Incoming trunk calls to the branch, as well as attendant-seeking voice-terminal calls, route over tie trunks to the main location.

Inter-PBX Attendant Service calls are incoming tie-trunk calls from a branch location to the main-location attendant group. If no attendant in the group is immediately available, the calls are queued. When an attendant becomes available, the call routes to that attendant. Extended calls are treated as incoming calls to the main location.

An Avaya switch can be a branch or main location. Users at each branch can access other branch locations through the main location. A branch can have local attendants. Users access these local attendants normally.

Administering Inter-PBX Attendant Service

Screen	Field
Tie Trunk Group (Main) (page 1)	● Incoming Destination
Console Parameters (Branch) (page 1)	● IAS (Branch) ● IAS Tie Trunk Group No. ● IAS Att. Access Code
Tie trunk group (Branch)	● All

About Inter-PBX Attendant Service interactions

- Attendant Control of Trunk-Group Access

  If a call at a branch attempts to access a controlled trunk group, the call routes to a branch attendant, if there is one. If there is no branch attendant, the call routes to the attendant group at the main location.

- Attendant Display and DCS Attendant Display

  In a DCS environment, an incoming call from a branch displays at the attendant console at the main location as a local call.

  In a non-DCS environment, an incoming call displays at the attendant console at the main location as an incoming tie-trunk call.

- Attendant Recall

  If an attendant at the main location holds a call, the calling parties at the branch cannot recall the attendant.

- Call Coverage

  A call redirected to a coverage path with the attendant group as a coverage point skips that coverage point. It goes to the next coverage point at the branch, if administered, or continues to ring at the previous coverage point. If the attendant group 0 is the only coverage point, it continues to ring at the principal's extension.

- Centralized Attendant Service

  CAS and Inter-PBX attendant calling cannot be used at the same time.

- Dial Access to Attendant

  Administer Dial Access to Attendant using the dial platform to the same digit on both the IAS main switch and the IAS branch switch. On the branch switch, administer the PBX attendant access code (**Console Parameters** screen) to match the main PBX attendant-group dial access code.

- Night Service

  Inter-PBX Attendant Calls deactivates when a branch goes into night service, and reactivates when the branch comes out of night service.

ISDN Feature Plus

The main ISDN Feature Plus topics included in this section are

- What is ISDN Feature Plus
- Administering ISDN Feature Plus
- About interrogation between message center and served user switches
- About ISDN Feature Plus interactions

What is ISDN Feature Plus

ISDN Feature Plus is an international feature, and does not apply to systems in the U.S. This feature allows you to have basic feature transparency over public networks without having a dedicated leased line. This provides a lower cost option using the switched public network.

ISDN Feature Plus uses Communication Manager proprietary signaling protocol. The features do not function in the same way as their QSIG or DCS counterparts.

To use Feature Plus, Phase I, you need Direct Inward Dialing (DID) extensions. In addition to the general Feature Plus call handling,

Feature Plus includes the following:

- Centralized AUDIX — A simple, one step “coverage” to voice mail. If voice mail is unavailable for any reason, the call does not cover elsewhere.
- Call Diversion — You can divert (or forward) calls unconditionally, upon busy or no reply, to another extension including forwarding voice mail.
- Calling Number ID — You can display the calling party’s number to the called party during alerting and after answer.
- Calling Name — You can assign the Calling Name Feature Plus identifier with a maximum size of 15 bytes or the maximum network subaddress size, whichever is lower.
- Connected Line Identification Presentation (COLP) — You can assign display forwarded-to party information to the calling user’s display.
- Call Transfer - Basic — You can transfer calls between parties across the public network. Display updates at the time of transfer or upon completion of transfer, however, are not supported.
- Served User PBX for Centralized AUDIX — Determines where to send messages destined for the AUDIX hunt group.
- Message Waiting Indication — You can display a message waiting indication on a user’s voice terminal.

Administering ISDN Feature Plus

Note:

Starting with Release 10, the system software release, Offer Category, features, and system capacities are controlled through the License File. The *init* login does not have the ability to change the customer options, offer options, and special applications screens. However, these screens are still available through the **`display system-parameters customer-options`** command.

To administer ISDN feature plus:

1. On the **System-Parameters Customer-Options** screen, verify that the:
   - **G3 Version** is *V7* or higher (page 1).
   - **ISDN Feature Plus** field is set to *y* (page 4).
2. On the same page, verify either one or both of the following:
   - **ISDN-PRI** field is set to *y*, or
   - **ISDN-BRI Trunks** field is set to *y*.
3. Verify either one or both of the following:
   - ISDN-BRI **Trunk Group** — Verify the **Supplementary Service Protocol** field is set to *`f`*
   - ISDN-PRI **Trunk Group** — Verify the **Supplementary Service Protocol** field is set to *`f`*.
4. On the **Feature-Related System-Parameters** screen (page 8), set the **Feature Plus Ext** field to the local extension used to terminate Feature Plus signaling for ISDN Feature Plus.
5. On the **Hunt Group** screen (page 2), to add a centralized AUDIX system, set the **Message Center** field to *`fp-mwi`*.

To start Message Waiting Indication at the Message Center PBX:

1. On the **Feature-Related System-Parameters** screen (page 8), set the **MWI - Number of Digits per AUDIX Subscriber** field to the desired number.
2. On the **Processor Channel Assignment**, set the **Application** field to *`fp-mwi`*.
3. Type **`change isdn mwi-prefixes`** and press **Enter**.

   Administer the **Message Waiting Indication Subscriber Number Prefixes** screen.

To start the Calling Name feature:

1. On the ISDN-BRI or ISDN-PRI **Trunk Group** screen (whichever you are using), set the **Send Name** field to *y*.

Differences in Inserted Digits field

There is a difference in how the **Inserted Digits to form Complete Number** field on the **Message Waiting Indication Subscriber Number Prefixes** screen is used for QSIG and for Feature Plus. This difference is because of how the Feature Plus and QSIG-TSC platforms operate:

- For Feature Plus, the Feature Plus extension must be included in the **Inserted Digits to form Complete Number** field
- For QSIG, only the higher order digits need to be included. (In QSIG MWI, the subscriber number is appended to the inserted digits and the resulting number is used to route over a QSIG TSC.)

For example, Dallas is a Message Center PBX and Denver is a remote PBX:

- If Feature Plus is running between Dallas and Denver and the Feature Plus extension in Denver is 82000, the **Inserted Digits to form Complete Number** field administered in Dallas to get to Denver must be 3035382000.

  The **Routing Digits (AAR/ARS Access Code)** field also needs to be filled in appropriately.
- If QSIG is running between Denver and Dallas, the **Inserted Digits to form Complete Number** field must contain 30353.

  The **Routing Digits (AAR/ARS Access Code)** field also must be filled in appropriately.)

About interrogation between message center and served user switches

When performing an audit, the Served User switch sends a request towards the Message Center switch. As a Message Center PBX, the Avaya switch receives the request message, maps it into a MW STATUS REQUEST - SINGLE STATION message, and sends it to AUDIX on the BX.25 link. When the AUDIX system replies to the DEFINITY or Avaya system on the BX.25 link with a MW STATUS UPDATE, the Message Center switch sends the information on to the appropriate Served User switch.

- If it is a Message Center PBX, the MW STATUS UPDATE indicates whether there are any messages waiting, not how many messages are waiting, or what media types are these messages. If the MW STATUS UPDATE indicates that there are new messages, then the Message Center PBX sends a message telling the Served User PBX to activate the message waiting indication. Similarly, if the MW STATUS UPDATE indicates that there are no new messages, then the Message Center PBX sends a message telling the Served User PBX to deactivate the message waiting indication.
- If it is a Served User PBX, when the Served User PBX receives the result, it makes sure that the result received from the Message Center matches the state of the Served User’s light.

About ISDN Feature Plus interactions

- Automatic Circuit Assurance

  Automatic Circuit Assurance (including Referrals) is not activated for calls terminating at the Feature Plus extension.

- Distributed Communication System (DCS)

  Feature Plus signaling links do not support DCS.

- Feature Plus Centralized AUDIX

  - Calling Line Identification Presentation (CLIP)

    If the public network supports CLIP and the called user has subscribed to the service, calling party information is available to the called user when messages are retrieved.

  - Feature Plus Diversion

    Feature Plus Centralized AUDIX relies upon Feature Plus Diversion. When a call covers to AUDIX, it must invoke Feature Plus Diversion to identify the called party to AUDIX.

  - Feature Plus Message Waiting

    When a calling party leaves a message using Feature Plus Centralized AUDIX, Feature Plus Message Waiting engages and turns on that subscriber’s message waiting indicator.

- Feature Plus Forwarding (Diversion)

  - Calling Line Identification Presentation (CLIP)

    If the public network supports CLIP and the forwarded-to user has subscribed to the service, then calling party information is available to the forwarded-to user’s display.

  - Connection Line Identification Presentation (COLP)

    If the public network supports COLP and the calling user has subscribed to the service, then forwarded-to party information is available to the calling user’s display.

  - Feature Plus Centralized AUDIX

    Feature Plus Centralized AUDIX relies upon Feature Plus Diversion. Invoke Feature Plus Diversion first to enable the Centralized AUDIX feature.

  - Call Coverage

    - Terminating call has coverage active

      If a call is forwarded off-switch, and the terminating switch has call coverage activated and the criteria are met, the call will not go to the forwarding coverage path. It goes to the terminating coverage path.

    - Forwarding and Coverage

      If the last coverage point in the coverage path is a number that routes over an ISDN SSF trunk, no Feature Plus Diversion information passes to the coverage PBX.

  - Automatic Callback

    If automatic callback was activated before the called voice terminal user activated Call Forwarding over an ISDN SSF trunk, the callback call attempt is redirected to the forwarded-to party over the SSF trunk.

  - Call Park

    If a forwarded-to (diverted-to) extension user parks a call that has been forwarded from an ISDN SSF trunk, the call normally is parked on the forwarded-to extension, not on the forwarded-from (called user) of the ISDN SSF trunk.

- Feature Plus Message Waiting Indication

  - Audio Information Exchange (AUDIX)

    Feature Plus MWI depends on the presence of a Message Center. Whenever an Avaya switch acts as a Message Center switch, there is an interaction between the switch and the AUDIX system. The switch must be able to receive messages from the AUDIX system then, if applicable, send the appropriate Feature Plus MWI message to the network. Similarly, if the switch receives a Feature Plus MWI message, the switch translates the Feature Plus message into the appropriate AUDIX message and passes it to the AUDIX system.

    The only messages that Communication Manager handles are AUDIX messages along the BX.25 link. Feature Plus MWI can interwork with Basic AUDIX, including INTUITY AUDIX, and with DEFINITY AUDIX with the DCIU control link. Feature Plus MWI does not work with the DEFINITY AUDIX that emulates a DCP voice terminal or with versions of AUDIX that communicate to Avaya mode codes.

    Implementation requires that all users on a Served User switch use the same Feature Plus Message Center. Some of the served users can use a Feature Plus Message Center, while others use a local message center and/or a DCS Remote Message Center and/or a QSIG Message Center. However, some served users on a switch cannot use one Feature Plus Message Center while other served users on the same switch use a different Feature Plus Message Center.

  - Off-Premise Station

    Feature Plus MWI does not work with an off-premise station implemented with a DS1 circuit pack.

- QSIG

  Feature Plus signaling links do not support QSIG.

Centralized Voice Mail

The main Centralized Voice Mail topics included in this section are

- About centralized voice mail
- What are mode code centralized voice mail configuration requirements
- Administering Centralized Voice Mail

About centralized voice mail

You can use a single voice mail system to support multiple Avaya switches and Merlin Legend/ Magix systems in a network using mode code.

This capability is available for:

- Avaya system software (Release 8 or later)
- Merlin Legend R6.1or later
- Merlin Magix 1.0 or later

Voice mail systems that support these connections are:

- Intuity AUDIX R4.4 or higher running on a MAP5, with up to 18 ports
- Octel 100, with up to 16 ports

Features that are supported

- Calling party name/number sending/retrieval
- Message waiting light activation
- Remote coverage to voice mail
- Fax, as well as voice, mail

Features that are not supported

The following capabilities are not supported in Centralized Voice Mail through mode code:

- Most DCS feature transparency
- Centralized voice mail for a tandem switch (does not have a direct connection to the hub switch)
- Transfer into voice mail

What are mode code centralized voice mail configuration requirements

Centralized voice mail using mode code requires the following:

- An Avaya switch as the hub of the voice mail network, with the voice mail system directly connected to it.
- Direct ISDN PRI tandem trunk connections, using DS1 service between the hub and the switches the voice mail supports.

  The system uses the D-channel to transmit mode code signals to light message waiting lights on remote extensions.
- A uniform dial plan for all switches in the network, with a 4-digit plan if Merlin Legend/ Magix is part of the network.
- One and only one mailbox for each extension in the network.

Note:

DCS software, X.25 hardware, and CLAN hardware/software are not required for this type of network. Additionally, you cannot network switches simultaneously using both mode code and DCS.

Note:

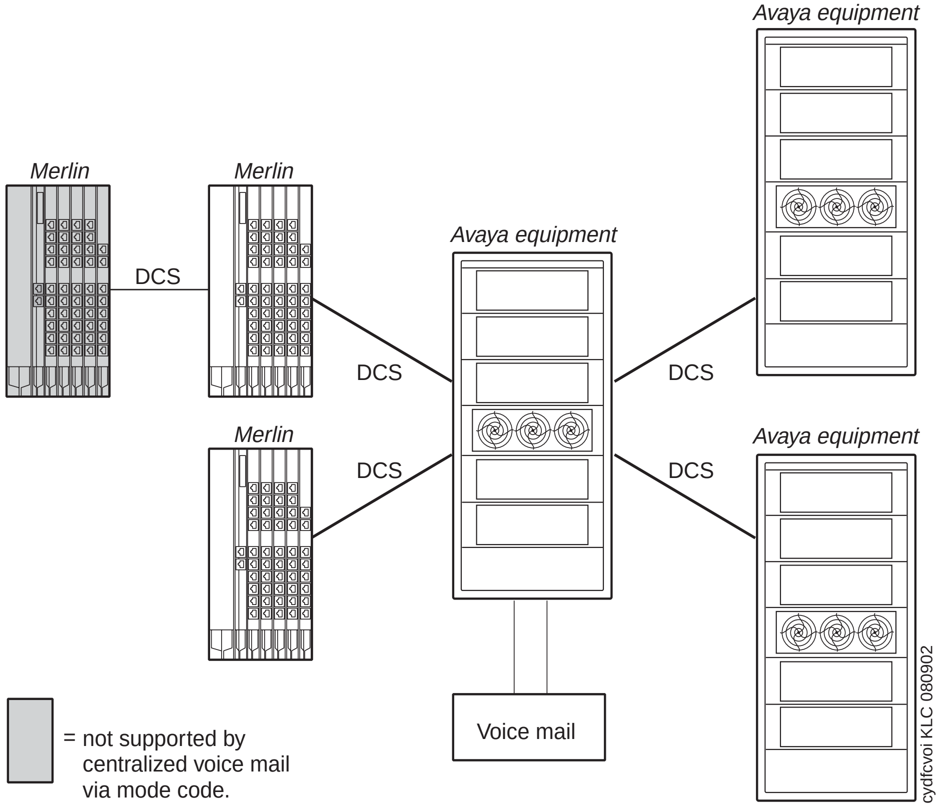
For Centralized Voice Mail using Mode Code, you’re network must be in a hub/ spoke configuration, with no more than ten DCS network nodes.

Centralized voice mail configuration using mode code example

Figure 13: Example of a Centralized Voice Mail configuration on page 228 shows what a configuration of centralized voice mail using mode code might look like.

In this configuration, system A is the hub. Voice mail system X is the centralized voice mail system. All other systems in the network are supported by voice mail system X *except* Legend system E and system D. These switches do not have a direct ISDN-PRI connection to the hub.

Figure 13: Example of a Centralized Voice Mail configuration



Administering Centralized Voice Mail

The following steps describe how to set up centralized voice mail. For information on setting up Merlin Legend/Merlin Magix, see your Merlin documentation. For information on setting up Intuity Messaging Solutions, see *Avaya IA770 INTUITY AUDIX Messaging, Release 2.0, Installation, Upgrades, and Troubleshooting*, 11-300399.

To administer centralized voice mail:

1. Enter **`display system-parameters customer-options`** on the SAT command line of your system administration screen.
2. On page 4, verify fields as follows:
   - **ISDN-PRI** field is `y.`
   - **Mode Code for Centralized Voice Mail** field is `y.`
   - Uniform dialing Plan (**UDP)** field is *y* (page 5).
   - **Mode Code Interface** field is *y* (page 6 of **System-Parameters Features** form).

3. On the hub switch, enter **`add trunk group`** ***`xxxx`*** on the command line of your system administration screen,

   where ***`xxxx`*** is the number of the ISDN-PRI trunk group connecting the hub with the remote switch.

4. On page 1, verify fields as follows:

   - **Group Type** field is `ISDN.`
   - **Service Type** field is *`TIE.`*

5. On page 2, verify fields as follows:

   - **Send Name** field is *y*.
   - **Send Calling Number** field is *y*.
   - **Format** field is *`Private`*.
   - **Send Connected Number** field is *y*.

6. On each remote switch, repeat steps 3-5.

7. On each switch in the network, enter **`change dialplan analysis`** on the SAT command line of your system administration screen.

8. Administer the dial plan for each node in the network.

   Usually the hub is considered Node 1.

9. For each node, enter **`change feature-access-codes`** on the command line.

10. On page 2, verify fields as follows:

    - **Leave Word Calling Send a Message** field is *`#90`*.
    - **Leave Word Calling Cancel a Message** field is *`#91`*.

    Note:

    All nodes in the system and the Voice Mail system must match this setting.

11. For each node, enter **`add ds1`** ***`UUCSS`*** on the command line, where ***`UUCSS`*** is the address of the DS1 circuit pack.

12. On page 1, verify fields as follows:

    - **Line Coding** field is *`B8ZS`*.
    - **Framing** field is *`extended superframe`*.
    - **Signaling Mode** field is *`isdn/pri`*.
    - **Connect** field is *`PBX`*.
    - **Interface** field is *`network`* (for the hub) and *`user`* (for the remote switch).

    Note:

    Mode Codes *will not work* with D4 or SuperFrame

13. For each node, enter **`change signaling-group`** ***`next`*** on the SAT command line.

    Administer the signaling group.

14. For each node, enter **`change private-numbering`**, and verify fields as follows:

    - **Set Network Level** field is *`0`*.

      This setting overrides the signaling on the D channel, allowing the Message Waiting lamp activation signal to be sent

15. On the Avaya node, enter **`change system-parameters mode-code`** on the SAT command line.

16. On the hub switch, set the **VMS Hunt Group Extension** field to the voice mail hunt group extension.

17. On the remote switches, repeat Step 15.

    Enter the voice mail hunt group extension in the **Remote VMS Extension - First** field.

18. For each node, enter **`change aar analysis`** on the SAT command line.

19. Verify the following:

    - **Call Type** field is *`lev0`*.

20. On the hub switch, enter **`change station`** ***`extension`*** for each port extension in the voice mail hunt group.

21. On Page 1, verify the following:

    - **Type** field is *`vmi`*.

22. On Page 2 of the **Station** screen, administer or verify the following:

    - **LWC Reception** field is *`msa-spe`* (Message Server Adjunct-System Processing Element).
    - **Leave Word Calling** field is *y*.
    - **Adjunct Supervision** field is *y*.
    - **Distinctive Audible Alert** field is *n*.
    - **Switchhook Flash** field is *y*.
    - **LWC Activation** field is *y*.

23. For each remote node, enter **`change coverage remote`** on the SAT command line.

24. Administer or verify the following:

    - **01** contains the extension of the voice mail hunt group.

Japan TTC Q931-a

The main Japan TTI Q931-a topics included in this section are:

- About Japan TTC Q931-a
- Considerations about TTC Basic Call Setup with Number Identification Supplementary Service
- What are the TTC Q931-a Protocols
- Administering Japan TTC Q931-a

About Japan TTC Q931-a

The Telecommunications Technology Committee (TTC) of Japan defines national standards that are to be used in domestic public and private network facilities. The TTC typically modifies other international standards as defined by ITU-T for use in Japan with additional national protocols to enhance operation for their customers.

The TTC has defined a family of Q931-a private networking protocols that allows for a level of feature transparency between different switches within a single vendor or multi-vendor private network. Communication Manager provides connectivity into the Japanese private networking environment through two methods:

- Channel Associated Signaling
- ISDN (Integrated Services Digital Network) PRI (Primary Rate Interface) — TTC specific protocol. Communication Manager supports Basic Call with Number Identification services.

Considerations about TTC Basic Call Setup with Number Identification Supplementary Service

Communication Manager allows the display of the calling party number to the called party. Communication Manager also displays the connected number to the calling party after the call connects to the called number of another destination. For many protocols, Number Identification is considered to be part of Basic Call service; however, the TTC protocol defines Number Identification services to be part of their supplementary services offering. No additional supplementary services are supported at this time.

You can administer outgoing calls as “yes”, “no”, or “restricted.” Restricted means that Communication Manager sends the information but sends it “presentation restricted,” which indicates to the receiving switch that the information should not be displayed. A received restricted number is included on the Call Detail Record (CDR), however.

What are the TTC Q931-a Protocols

The TTC defined private networking ISDN protocol is largely based upon the ITU-T Q.931 protocol. Communication Manager supports the following TTC defined protocols:

- Basic Call support as defined in JT-Q931-a “Digital Interface between PBXs (Common Channel Signaling) — Layer 3”
- Number Identification Services as defined in JT-Q951-a “Digital Interface between PBXs (Supplementary Services) — Number Identification Services”

Differences from ITU-T Q.931 include:

- Symmetrical operation as Peers similar to QSIG protocol, i.e. No Network/User definition.
- Different protocol discriminator.
- Progress Indicator IE not supported in DISCONNECT messages.
- Timers T310 and T313 are disabled.
- Sending Complete IE not supported.
- NOTIFY messages are not supported.

Administering Japan TTC Q931-a

To administer Japan TTC Q931-a connections:

1. Verify that you have the appropriate circuit pack for integration
2. Enter **`display system-parameters customer-options`** on the SAT command line.
3. On page 1, verify that the **G3 Version** field is *`V8`* or later
4. On page 4, verify that **ISDN-PRI** field is *y*.
5. Administer the TTC DS-1 circuit pack.

   Check for the following field entries:

   - **Connect** field — *`pbx`*
   - **Interface** — *`peer-master`* or *`peer-slave`*
   - **Peer Protocol** — *`TTC`*
   - **D-channel** — *must match between the local and receiving switches*)
   - **Channel Numbering** — *`sequential`* or *`timeslot`* (*This field must be the same on both the local and receiving switches*)

6. Administer or check the TTC ISDN trunk group(s) associated with the DS1 circuit pack.

   Check for the following field entries on page 1 of the **Trunk Group** screen:

   - **Group type** — *isdn*
   - **Supplementary Service protocol** — *a*
   - **Outgoing Display** — *y*
   - **Disconnect Supervision — In?__Out?__**

   Check for the following field entries on page 2 of the **Trunk Group** screen:

   - **Format** — *public*, *private*, *unknown*, *unk-pvt*
   - **Send Connected Number** — *y*
   - **Sending Calling Number** — *y*
   - **Send Name** — *n*

Chapter 5: Feature interactions and considerations

This appendix contains feature descriptions, considerations, and interactions for

- Distributed Communication System
- QSIG interactions
- Centralized Attendant Service (CAS) interactions and considerations
- Italian TGU/TGE (main and satellite) interactions
- Hairpinning and shuffling feature interactions

Distributed Communication System

Extension Number Portability considerations

- If you use DCS, the Extension Number Portability (ENP) node numbers must correspond to DCS node numbers.

DCS over ISDN-PRI D-channel considerations and interactions

Considerations

- The gateway node serves as the terminating node to the D-channel DCS network as well as the terminating node to the traditional DCS network.

  A switch serving as an ISDN DCS Gateway node introduces some interesting situations when administering processor channels in an associated traditional DCS system. In a traditional DCS network, (BX.25 processor channel links) Remote Port in the **Processor Channel Assignments** screen refers to the processor channel of the destination switch. In an Integrated DCS network, Remote Proc Chan in the **Processor Channel Assignments** screen refers to the processor channel of the Gateway switch (if the destination switch is an ISDN DCS system), *not* the destination switch.

  On the contrary, Machine-ID in the **Processor Channel Assignments** screen refers to the destination switch, either an ISDN DCS system or a traditional DCS system. The Gateway switch number must not be used in this field if the destination switch is an ISDN DCS system.

Interactions

- ASAI

  For incoming calls on DCS over ISDN-PRI, ASAI applications receive the ISDN-PRI Calling Party Information, not the DCS Calling Party Information.

- Attendant DXS with Busy Lamp Field

  An attempt by the attendant to select directly an extension that has been previously administered as belonging to an administered NCA-TSC results in the intercept tone being received.

- **CDR**

  CDR records both the status and the use of TSCs. Both CA-TSCs and NCA-TSCs can be recorded. For more information, consult the CDR description in this manual or the CDR manual.

- D-channel Backup

  In the event of a D-channel switchover (primary to secondary or secondary back to primary) in a private network, administered NCA-TSCs that were active are assumed to have remained active. Any unacknowledged user-user service requests are assumed to be rejected, and administered NCA-TSCs which were in the process of being established at the time of the switchover are dropped when the switchover occurs. Those administered NCA-TSCs that were dropped are reattempted again.

  If a D-channel switchover occurs on a D-channel going to the public network then all TSCs are dropped. A maintenance-provided “heartbeat” message periodically is sent over each permanent administered NCA-TSC to ensure that such a situation is detected and rectified.

- Distributed Communications System AUDIX (DCS AUDIX)

  The DCS over ISDN-PRI D-channel feature can be used to support DCS AUDIX. (The connection between si and the AUDIX system should be BX.25 or CLAN.)

- **GRS**

  GRS selects TSC compatible facilities when routing NCA-TSCs. In other words, a NCA-TSC request can only select a routing preference that supports TSCs.

  In a tandem node, GRS first selects facilities that support TSCs if the call falls into any one of the following two conditions:

  - It requests a CA-TSC explicitly
  - It contains a DCS information element in the SETUP message

  Once a trunk group with available members is selected, the call proceeds even if all the TSCs belonging to the associated signaling group are active. In other words, the completion of a call is given priority over DCS transparency.

- AT&T SDN or MCI N-Quest

  The DCS over ISDN-PRI (DCS+) D-channel feature allows the system to access public networks, such as AT&T SDN or MCI N-Quest. DCS+ supports all DCS features except for the following:

  - DCS Attendant Control of Trunk Group Access
  - DCS Attendant Direct Trunk Group Selection
  - DCS Busy Verification of Terminals and Trunks

- Voice Terminals

  An attempt to dial an extension that has been previously administered as belonging to an administered NCA-TSC results in the intercept tone being received.

Enhanced DCS considerations and interactions

Considerations

- If the DCS link fails, the administrator can choose to allow calls to continue without class of restriction checking or to block all DCS calls to inward-restricted stations.

Interactions

- Class of Restriction

  When a call goes to coverage, it is the called party’s (not the covering party’s) restrictions that are used.

DCS feature descriptions, interactions and considerations

Table 33: DCS feature descriptions, interactions, and considerations on page 238 lists DCS features, gives a brief description, and describes the feature interactions and considerations for their use.

Table 33: DCS feature descriptions, interactions, and considerations *1 of 9*

Feature	Description	Interactions	Considerations
Alphanumeric Display for Terminals	This feature allows calling-name display, called-name display, and miscellaneous identifiers to be transferred from a terminal on one node to a terminal on another node.	See Alphanumeric Display interactions	On outgoing DCS calls, display of the called name may be delayed for a few seconds until the required information arrives from the distant node. The called name display only works between Avaya switches
Attendant Control of Trunk Group Access	DCS Attendant Control of Trunk Group Access allows an attendant at any node in the DCS to control an outgoing trunk group at an adjacent node in the cluster. The attendant uses a remote-tgs feature button on the console for this purpose. To use this feature, you must have a DCS Trunk Group between the local and remote switches, and the trunks in that trunk group cannot insert digits on incoming calls. If you need digit insertion on these trunks, it should be added on the outgoing trunk based on the dialed string. **NOTE:** DCS Attendant Control of Trunk Group Access is not available if you are using D-channel DCS.	● DCS Attendant Display ● When a user attempts to access a controlled trunk group and is routed to the local attendant, the display shows the reason the call was redirected. If the call is routed using CAS or the Inter-switch Attendant Calls feature, the display does not show the reason the call was redirected. ● UDP ● DCS tie trunks should not be attendant controlled. This would result in all UDP calls on the controlled tie trunk being routed to the controlling attendant instead of to the desired destination.	● This feature is not available for trunk groups with 4-digit trunk access codes or for trunk members 100 through 999. ● If the remote node (where the trunk group to be controlled resides) is a System 75, Generic 1, or Generic 3, it is not necessary for that node to have an attendant console with corresponding three-lamp Trunk Hundreds Select button. However, if the remote node is a System 85, Generic 2.1, or Enhanced DIMENSION system, control of the trunk group is not allowed unless an attendant at that node has a corresponding three-lamp Trunk Group Select button. ● The attendant must use the Remote Trunk Hundreds Select button to directly access the controlled remote trunk group. If an attendant controls a remote trunk group, and that attendant dials the trunk access codes of the DCS tie trunk and the controlled remote trunk group, the call is routed to the attendant at the node where the trunk group resides. ● If Attendant Control of Trunk Group Access is activated, and no attendant is assigned, or the attendant is later removed, calls to a controlled trunk group route to the attendant queue.

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Table 33: DCS feature descriptions, interactions, and considerations *2 of 9*

Feature	Description	Interactions	Considerations
Attendant Direct Trunk Group Selection	DCS Attendant Direct Trunk Group Selection allows attendants at one node to have direct access to an idle outgoing trunk at a different node in the DCS. This feature functions the same as regular Direct Trunk Group Selection. However, the attendant uses a remote-tgs feature button on the console for this purpose. **NOTE:** DCS Attendant Direct Trunk Group Selection is not available if you are using D-channel DCS. To use this feature, you must have a DCS Trunk Group between the local and remote switches, and the trunks in that trunk group cannot insert digits on incoming calls. If you need digit insertion on these trunks, it should be added on the outgoing trunk based on the dialed digits. You can assign a Trunk Hundreds Select button to access a trunk group at the local node or a trunk group at a remote node. A Trunk Group Select button assigned to access a remote node is referred to as a remote Trunk Hundreds Select button. Pressing a remote Trunk Group Select button is the same as dialing the tie trunk group access code for the remote node and the trunk access code of the selected trunk.		This feature is not available for trunk groups with 4-digit trunk access codes or for trunk members 100 through 999.
			2 of 9

Table 33: DCS feature descriptions, interactions, and considerations *3 of 9*

Feature	Description	Interactions	Considerations
Attendant Display	The DCS attendant console displays calling-party ID and called-party ID information for calls to and from remote switches in the network.	When both ISDN and DCS display information, or only DCS display information, is received, the switch displays the DCS display information in the DCS format. If ISDN display information is received, and no DCS display information is received, then the ISDN display information displays in the ISDN formats.	If you call an attendant on another switch in the DCS network, your display shows the attendant's name, but does not show the attendant's extension, instead you see a zero where the extension should be. CORs for an Avaya switch might not correspond to those used by an Enhanced DIMENSION system, System 85, or DEFINITY system Generic 2.1. Therefore, if the DCS network contains nodes other than Generic 1 or Generic 3, the display CORs may be misinterpreted. If it is important that certain CORs between various systems correspond with each other, those CORs should be administered accordingly. On outgoing calls, the display of called party information may be delayed a few seconds until the required information arrives from the remote node. The called party information is displayed only if both nodes are Generic 1 or System 75. DCS tie trunks between nodes must be administered with the Outgoing Display enabled. This enables the called party's name to be displayed at the calling attendant's display.
			3 of 9

Table 33: DCS feature descriptions, interactions, and considerations *4 of 9*

Feature	Description	Interactions	Considerations
Automatic Callback	DCS Automatic Callback allows a user at one node to make an automatic callback call to a user at another node in the DCS. A DCS Automatic Callback call can be initiated from a terminal at one node to a terminal at another node in the same way as if at a local node under the following conditions. ● If the called party is at a System 85, Generic 2, or Enhanced DIMENSION PBX node, the callback call can only be activated if the called node is returning busy tone or special audible ringback. ● If the called party is at a Generic 3, Generic 1 or System 75 node, the callback call can be activated if the called node is returning busy tone, Call Waiting ringback tone, or ringback tone. ● The calling party must disconnect within 6 seconds after hearing the confirmation tone for Automatic Callback activation. **NOTE:** If the calling party is on a System 85, Generic 2, or Enhanced DIMENSION PBX node and is unable to receive the callback call (for example, a busy single-line voice terminal without Call Waiting), Automatic Callback is reactivated by the calling party's node. If the calling party is on a Generic 3, Generic 1, or System 75 node and is unable to receive the callback call, the callback call is canceled.	Attendant Control of Trunk Group Access and DCS Attendant Control of Trunk Group Access Automatic Callback cannot be activated if the call uses a controlled trunk group.	An Automatic Callback request is canceled automatically if the called party does not become available within 40 minutes, or if the calling party does not hang up within six seconds after activating Automatic Callback.
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Table 33: DCS feature descriptions, interactions, and considerations *5 of 9*

Feature	Description	Interactions	Considerations
Automatic Circuit Assurance	DCS Automatic Circuit Assurance (ACA) allows a voice-terminal user or attendant at a node to activate and deactivate ACA referral calls for the entire DCS network. This transparency allows the referral calls to originate at a node other than the node that detects the problem. If referral calls are generated at a node for one or more remote nodes, the remote nodes are notified when ACA referral is activated or deactivated.		
Busy Verification of Terminals and Trunks	DCS Busy Verification of Terminals and Trunks allows attendants and multi-appearance voice-terminal users to make test calls to voice terminals and trunk groups that are located at other nodes in the DCS. To use this feature, you must have a DCS Trunk Group between the local and remote switches, and the trunks in that trunk group cannot insert digits on incoming calls. If you need digit insertion on these trunks, it should be added on the outgoing trunk based on the dialed digits. Multi-appearance voice terminal users can busy-verify an adjunct at a remote location by pressing Verify and dialing the TAC of the tie trunk group to the remote node. Then they must press Verify a second time and dial the desired TAC and the trunk group member number to be verified. Verification of the trunk then continues as if the trunk is on the same node.	If the Trunk Identification by Attendant feature is used during busy verification of a trunk (Trunk ID button is pressed), the trunk access code and trunk group member number of the DCS tie trunk being used is displayed. DCS Busy Verification of Terminals and Trunks transparency is lost if the routing pattern is administered to not delete the RNX and the AAR prefix is inserted on the terminating switch trunk group. The voice terminal display at the terminating switch displays only **a=station name**. **Extension** is left blank.	
Call Coverage	See Call Coverage considerations	See Call Coverage interactions	
			5 of 9

Table 33: DCS feature descriptions, interactions, and considerations *6 of 9*

Feature	Description	Interactions	Considerations
Call Forwarding	DCS Call Forwarding allows all calls to an extension to be forwarded to a selected extension in the DCS network or to an external (off-premises) number. If the Call Forwarding and DCS Call Forwarding are both active, and if a call is forwarded between extensions on the same node, the Call Forwarding coverage path is used. If the nodes are different, the DCS Call Forwarding coverage path is used. Voice-terminal users in the DCS can activate/deactivate this feature with a dial access code or with a Call Forwarding button. **NOTE:** Calls can be forwarded to a Vector Directory Number (VDN) anywhere in the DCS network. An attendant cannot activate/deactivate Call Forwarding for a VDN.	If the forwarding extension and the designated extension are at different nodes, and the designated extension's coverage criteria are met on a forwarded call, the call is redirected to a point in the designated extension's coverage path. If the forwarding extension and the designated extension are at different nodes, LWC and Coverage Callback cannot be activated at the designated extension for a forwarded call. There is a 30-second interval during which calls forwarded from the switch to another DCS node is denied. This prevents forwarded incoming trunk calls from being forwarded ad infinitum between two extensions.	
Call Waiting	DCS Call Waiting allows calls from one node to busy single-line voice terminals at another node to wait until the called party is available to accept the call. With DCS Call Waiting, a single-line voice terminal user, by knowing a call is waiting, can quickly process calls from locations within the DCS. DCS Call Waiting functions the same as normal Call Waiting. DCS Call Waiting includes the following features: ● Attendant Call Waiting ● Call Waiting — Termination ● Priority Calling DCS priority calling from the attendant station is *not* available.	DCS Call Waiting is denied when the following features are activated at the single-line voice terminal: ● Automatic Callback (to or from the voice terminal) ● Data Privacy ● Data Restriction On incoming trunk calls to the attendant extended over DCS trunks, Attendant Call Waiting interacts with the EDCS feature.	
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Table 33: DCS feature descriptions, interactions, and considerations *7 of 9*

Feature	Description	Interactions	Considerations
Distinctive Ringing	DCS Distinctive Ringing activates the called-terminal alerting or ringing device to indicate the type of incoming call to the user before they answer it. Distinctive Alerting functions in a DCS environment the same as it does within a single system. By default, internal calls are identified by a1-burst ringing pattern, external calls by a 2-burst ringing pattern, and priority calls by a 3-burst ringing pattern. However, you can administer these patterns.	Distinctive Ringing treats a call from another switch in a DCS arrangement as external; DCS Distinctive Ringing treats such calls as internal. If both features are administered, DCS Distinctive Ringing takes precedence. If EDCS is activated, DID treatment may be different. See DCS+ configurations on page 169.	
Leave Word Calling	LWC transparency in a DCS configuration allows messages from an Avaya switch to another node, depending on the storage capability of the remote node.	DCS Multi-appearance Conference/Transfer Activation of LWC is denied after a DCS call has been conferenced or transferred.	LWC cannot be successfully activated toward any system that is not capable of storing the messages, either internally or in an associated adjunct. Messages from one node, through an intermediate node, to a remote node do not require storage capability at the intermediate node. LWC transparency is supported for all DCS configurations except for cases when either the activating node or the remote node is either an ENHANCED DIMENSION system or a System 85 R2V1. Retrieval of LWC messages is permitted only from a terminal at the node where the messages are stored. DCS LWC cannot be activated from an attendant console.
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Table 33: DCS feature descriptions, interactions, and considerations ***8 of 9***

Feature	Description	Interactions	Considerations
Multi-appearance Conference/ Transfer	DCS Multi-appearance Conference/Transfer provides transparency for transferring calls and conferencing calls within a DCS network. A user in the DCS can initiate conference calls among or transfer calls originated from extensions in the DCS network to another extension within the DCS by dialing the UDP extension. (For transferred calls, the destination need not be within the DCS.) In a DCS, if a party in a conference hangs up or completes a transfer leaving only outgoing trunks on the call, the system attempts to preserve the connection if any of the remaining parties on the call is a DCS tie trunk.	Voice Terminal Display No display transparency is provided for DCS Multi-Appearance Conference/Transfer. EDCS On calls to or from Public Network Trunks, calling/ called party restrictions are checked when EDCS is active.	
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Table 33: DCS feature descriptions, interactions, and considerations *9 of 9*

Feature	Description	Interactions	Considerations
Trunk Group Busy/Warning Indication	DCS Trunk Group Busy/ Warning Indication provides attendants with a visual indication that the number of busy trunks in a remote group reached an administered level. A visual indication is also provided when all trunks in a trunk group are busy. **NOTE:** DCS Trunk Group Busy/Warning Indication is not available if you are using DCS over ISDN-PRI. To use this feature, you must have a DCS Trunk Group between the local and remote switches, and the trunks in that trunk group cannot insert digits on incoming calls. If you need digit insertion on these trunks, it should be added on the outgoing trunk based on the dialed digits. Except for legacy System 75, System 85, and DEFINITY G2 switches, you can administer DCS Trunk Group Busy/Warning Indication only for remote trunk groups that are directly connected to the local switch. Trunk group access codes for these trunk groups must be 3 digits or less and cannot include trunk members 100 through 999.	Loudspeaker Paging Access If Trunk Hundreds Select buttons are assigned for Loudspeaker Paging Access zones, Trunk Group Busy Indicators provide a visual indication of the busy or idle status of the zones at the remote location as well as at the local node.	Trunk Group Busy and Trunk Group Warning Indication is particularly useful with the Attendant Control of Trunk Group Access feature. The indicators alert the attendant when control of access to local and remote trunk groups is necessary.
DCS with Rerouting	See DCS with Rerouting considerations		
			9 of 9

Call Coverage considerations

DCS Call Coverage provides DCS messaging required for calls to be covered on remote systems when there is a DCS signaling link (BX.25, PPP, or ISDN-PRI) for the trunk groups. Calls to an extension on one system are covered by extensions on remote systems that are administered as coverage points.

Figure 14: DCS Call Coverage on page 247 shows an example of DCS Call Coverage.

Figure 14: DCS Call Coverage

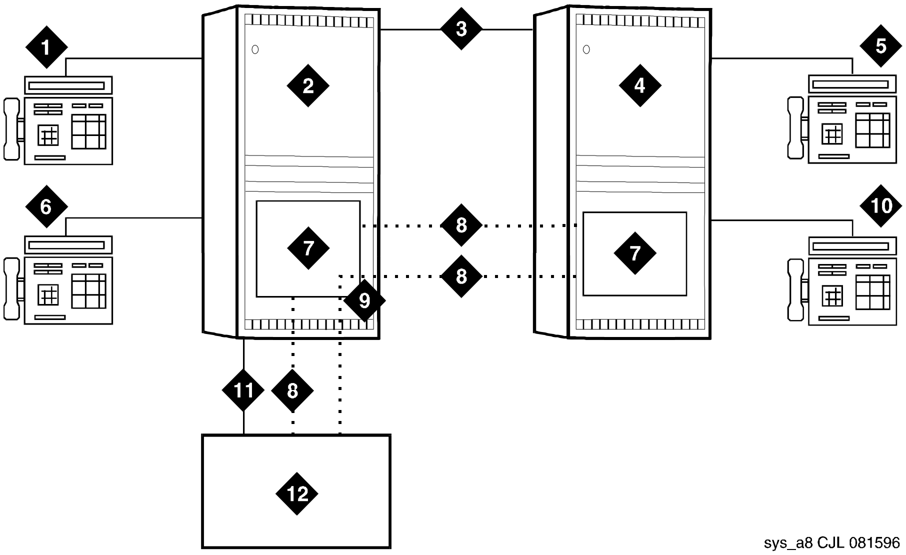

Figure notes:

1. **Station A**
2. **Avaya System A**
3. **DCS Tie Trunk Groups**
4. **Avaya System B**
5. **Station C**
6. **Station B**
7. **PGATE or PI Board**
8. **X.25 or ISDN PRI DCS Signaling Link**
9. **Hop or ISDN TSC Gateway**
10. **Station D**
11. **AUDIX Voice Lines**
12. **AUDIX - x34000**

In the figure, calls to Station A can be covered first by Station B, then by Station C or D, and finally by the AUDIX system connected to system A. Alternatively, calls could be covered by Station C, then Station B, then Station D, and so on.

If the called party answers after the call goes to coverage and the coverage point has answered, then the called party, calling party, and coverage point are all conferenced together.

If the called party answers and the coverage point has not answered, the call to the coverage point drops and the called party connects to the calling party.

Exceptions to DCS Call Coverage

DCS Call Coverage is similar to Call Coverage, with the following exceptions:

- Coverage Answer Groups across nodes are not supported.
- Under the following error conditions, a call follows the coverage point's coverage path.

Error Condition	Action
DCS link not up. or DCS trunk is not available. or DCS Call Coverage feature is not activated on the remote system.	The call is routed to the remote coverage point. If the call is answered, it is treated as Call Coverage Off Premises (also called Remote Call Coverage). If the call is redirected at the remote coverage point before the DCS SRI expires, the remote point's path is followed. If the call is not answered within the DCS SRI time-out period, the next coverage point is tried with DCS Call Coverage from the local system.
All trunks to the remote system, DCS or otherwise, are busy	The next coverage point is tried with DCS Call Coverage from the local system.

- When the DCS link is down, call consult operates differently. If Station A calls Station B but the call covers to Station C, then Station C consults back to Station B and Station B receives the consult call on the next call appearance.
- DCS Call Coverage does not support Coverage Call Back from a remote node.

Additionally, in some DCS Call Coverage situations, call coverage operation may deviate, including:

- A call to the principal redirects to the remote coverage point, which is unavailable. The coverage point is considered unavailable when:
  - The coverage point is not a valid extension, QDN, or VDN.
  - The coverage point is busy with no hunting, forwarded, or has send all calls activated, or activates send all calls after ringing.
  - The coverage point has no staffed agents or an invalid vector.

    When the coverage point is unavailable, the local system determines the availability status from a time-out or from a message from the remote system. When the local system discovers that the coverage point is unavailable, it tries the next coverage point. If the last coverage point is unavailable, the previous coverage point rings until it is answered or until the caller hangs up. If only one coverage point exists in the path and it is unavailable, the principal's station rings until it is answered or until the caller hangs up.
- A call to the principal is forwarded and the forwarded-to extension is not available. In this case, the first coverage point in the principal's path is tried. Note that the coverage does not follow the forwarded-to extension's coverage path.

- A call to the principal redirects to the remote coverage point, which answers. Subsequently, the principal goes off hook. In this case, the local system bridges the principal onto the call between the calling party and coverage point creating a conference among the three. The principal receives the call on the same call appearance as the original call.
- A call to the principal redirects to the remote coverage point. While the remote coverage point is ringing, the principal answers the call. In this case the call is not cut through to the coverage point. Instead, ringing and ringback is removed from the coverage point and the call is cut through to the principal.

DCS with Rerouting considerations

DCS with Rerouting allows a call's connection between two Avaya switches to be replaced by a new connection. All of the trunks used in the original path must be DCS+ (DCS over PRI) and the new path utilizes only DCS+ trunks. DCS with Rerouting provides the following capabilities:

- Attempts to obtain a better (generally less expensive) connection.
- May replace the current path of a call with a route that is better in terms of Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) routing preferences administered on an Avaya switch.
- Frees up resources being used unnecessarily.

DCS with Rerouting must be enabled on a switch-wide basis and the trunk groups involved must be administered as SSE. DCS with Rerouting primarily provides you with the ability to be more effective with the usage of Trunk groups administered for Supplementary Services Protocol Option E (SSE) during the existence of an active call. This means using a preferred route (in terms of UDP/AAR/ARS routing preferences administered on the switch) between the switches involved.

DCS with Rerouting can be invoked after Call Coverage. This Call Coverage also applies to AUDIX calls.

To invoke DCS with Rerouting:

1. On page 1 of the **System-Parameters Call Coverage/Call Forwarding** screen, enter **n** in the **Maintain SBA at Principal** field.

   DCS with rerouting can only occur if you *do not* need to maintain a simulated bridged appearance at the principal.

2. On page 3 of the **System-Parameters Customer-Options** screen, verify **DCS with Rerouting** field set to *y*.

3. On page 1 of the **Trunk Group** screen, enter **e** in the **Supplementary Services Protocol** field.

   This option allows limited QSIG signaling over DCS trunks.

   To enable this value (**e**), review the following fields on this screen for the appropriate values:

   - **DCS with Rerouting** must be set to *y*.
   - **Service Type** must not be set to *`dmi_mos`* or *`SDDN`*.

4. On page 2 of the **Trunk Group** screen, review the following fields for the appropriate values:

   - **Used for DCS** must be set to *y*.
   - **Send Name** can only be set to **y** or **n**. You cannot use **restricted**.

Users can invoke DCS with Rerouting by **Call Transfer**, **Transfer out of AUDIX**, and **dial 0 out of AUDIX**.

Alphanumeric Display interactions

The following features allow transparency with respect to Calling or Called Name Display and miscellaneous ID.

- Call Coverage

  At the calling terminal, the miscellaneous id “cover” is not displayed.

- Call Forwarding

  When a system user calls a party on a different node in the DCS and the call is forwarded, the miscellaneous ID “forward” is not displayed. At the covering (forwarded-to) user’s terminal, only the calling party’s name is shown; the called party’s name is not displayed.

- Call Park

  When a DCS call between a local system user and a user on another node is parked by the remote user, the miscellaneous ID “park” is not displayed at the local terminal.

- Call Pickup

  When a DCS call from a system user to another node is answered by way of Call Pickup, the miscellaneous ID “cover” is not displayed at the caller’s terminal.

- Call Waiting

  When a DCS call from a system user to another node is waiting at the called terminal, the miscellaneous ID “wait” is not displayed at the caller’s terminal.

- **CAS**

  When a user dials the extension for CAS, a RLT is seized or the caller is queued for an RLT. The caller’s terminal displays the trunk group identifier, such as OPERATOR.

- ISDN-PRI

  If both DCS and ISDN-PRI features are provided with a system, the ISDN-PRI display information displays in DCS format.

Call Coverage interactions

DCS Call Coverage has the same interactions as Call Coverage plus the following additional interactions:

- Call Coverage Off Premises

  If the coverage point is a non-UDP number in the remote call coverage table, Call Coverage Off Premises is applied to the call rather than DCS Call Coverage, even if a DCS link exists to the remote system.

- Coverage Answer Groups

  DCS Call Coverage to Coverage Answer Groups on remote systems are not supported by DCS Call Coverage. Coverage answer groups cannot be administered on a system other than the principal's system.

- Coverage Call Back

  DCS Call Coverage does not support Coverage Call Back from a remote node.

- Displays

  The displays on the DCS Call Coverage point's terminal may be different than those associated with the Call Coverage feature in the following situations:

  - When the call from the calling party to the principal or the redirected call to the coverage point travel over ISDN-PRI trunk groups.
  - When the calling party is on a System 85 or Generic 2.
  - When the DCS name message is not received by the remote (coverage point's) system.

- Go to Cover

  Go to Cover is not supported over DCS and therefore is not supported with DCS Call Coverage.

- Leave Word Calling Back to Principal

  With DCS Call Coverage, a covering user on a different node cannot press their LWC button to leave a message for the principal to call the covering user.

- Queuing

  DCS Call Coverage interacts with queuing in the following way. If a call is queued to a coverage point, such as a queue to a hunt group or an ACD split, and the queue is not full, the call remains in the queue without subsequent redirection until answered or until the caller hangs up.

DCS with Rerouting interactions

- When interworking with non-ISDN trunks or non-Supplementary Service Option E ISDN trunks, the system acts as a gateway in the following sense:
  - When a call is tandeming through an Avaya switch from a non-ISDN trunk to an SSE trunk or from a non-Option E to an SSE trunk, the system acts as an incoming gateway.
  - When a call is tandeming through an Avaya switch from an SSE trunk to a non-ISDN trunk or from an SSE trunk to a non-Option E trunk, the system acts as an outgoing gateway.

    As an example, when calls come in from the public network to the DCS network and then are transferred to another extension within the private network, the Avaya switch functions as an incoming gateway and rerouting occurs.
- If a conference call is transferred, rerouting will not occur.

QSIG interactions

QSIG interactions included in this section are

- QSIG/DCS Interworking
- Call Forwarding (Diversion)
- Call Transfer
- Transfer Into QSIG Voice Mail
- QSIG Name and Number Identification
- Path Replacement (PR)
- Transit Counter (TC)
- Call Completion (CC)
- Message Waiting Indicators (MWI)
- Called/Busy Name and Number
- VALU Call Coverage
- QSIG Centralized Attendant Service

QSIG/DCS Interworking

No features are interworking between QSIG and traditional DCS. With DCS+, only the following features are interworking:

- Name and number transport
- Voicemail
- Leave word calling

Call Forwarding (Diversion)

The interactions that apply to the standard Call Forwarding features also apply to Call Forwarding (Diversion) with QSIG. The following are additional interactions.

- Alternate Facilities Restriction Levels

  The AFRL of the original call is the AFRL used for Call Forwarding with Reroute.

- Authorization Codes

  Call Forwarding with Reroute is denied to calls that require an Authorization Code.

- Automatic Alternate Routing and Automatic Route Selection

  Call Forwarding with Reroute uses AAR and ARS to reroute the original call.

- Call Detail Recording

  Call Forwarding with Reroute is denied to calls that require Forced Entry of Account Codes.

- Call Transfer

  When a forwarded call transfers, the forwarding indication displays to the caller until the call is answered. This display includes the trunk group name and word “forward.” When the call is answered, the word “forward” is removed and the name and number of the answering party displays.

- Distributed Communications Systems

  Call Forwarding feature transparency does not exist on calls tandemed between a QSIG (Supplementary Service protocol b) network and a traditional DCS network. However, the basic call continues.

- Facility Restriction Levels and Traveling Class Marks

  The FRL (and TCM) of the original call is the FRL used for Call Forwarding with Reroute.

- Forwarding and Coverage

  If a coverage point is a number that routes over an ISDN (Supplementary Service protocol b) trunk, QSIG diversion information is passed to the coverage switch.

- QSIG Name and Number Identification

  Availability of name and/or number display at the originating and diverted-to users depends upon how QSIG Name and Number Identification has been administered for the switches involved.

- Terminating Call has Coverage Active

  If a call is forwarded off switch, and **Cover after Forward** is set to *y* on the **Feature-Related System-Parameters** screen, then the call will follow the original called party’s cover path. If the **Cover after Forward** field is set to *n*, the terminating switch has call coverage activated, and the criteria are met, the call does not route to the forwarding party’s coverage path. It routes to the terminating station’s coverage path.

Call Transfer

- Call Forwarding (Diversion)

  When a call is forwarded and transferred or transferred and forwarded, the forwarding indication displays to the caller until the call is answered. This display includes the trunk group name and word “forward.” When the call is answered, the word “forward” is removed and the name and number of the answering party displays.

- Distributed Communications Systems

  The only DCS transparency that exists when a call is transferred in a DCS network and passed over a QSIG administered trunk is calling name and number.

- QSIG Path Replacement

  PR is invoked whenever a QSIG transferred call is answered.

- QSIG Name and Number Identification

  Availability of name and/or number display at the connected parties depends upon how QSIG Name and Number Identification has been administered for the switches involved.

Transfer Into QSIG Voice Mail

- QSIG Path Replacement

  After a call is transferred into QSIG voice mail and the voice mail system answers the call, Path Replacement is attempted.

QSIG Name and Number Identification

- Distributed Communications Systems (DCS+)

  In a DCS+ network, Communication Manager displays the DCS called name/number information or it will display ISDN connected name/number information, depending upon who answers the call.

  When an incoming ISDN call is routed back out over a non-ISDN trunk group, Communication Manager can send the name of the non-ISDN trunk group as the connected name if the **Send Non-ISDN Trunk Group Name as Connected Name** field is *y* on the **Feature-Related System-Parameters** screen.

  Communication Manager interworks called/calling/connected name and number identification between DCS+ and QSIG.

Path Replacement (PR)

- Basic Call Management System

  On a connection monitored by a BCMS entity, PR is allowed.

- Call Detail Recording

  Codes for recording the new connections of PR calls are code J for incoming trunk calls and code K for outgoing trunk calls. When a path is replaced, you also may receive records for short-duration calls that are not directly linked to the J and K records.

- Call Management System

  On a connection monitored by a CMS entity, PR is allowed.

  Note:

  > Communication Manager sends updates for transfer and conference to BCMS and CMS to make reports complete. Path Replacement is allowed.

- Call Vectoring

  A transferred call that terminates at a vector and is answered can have its path replaced.

- Data-Call Setup

  A data call is denied PR.

- Data Privacy

  If Data Privacy is active, PR is denied.

- Data Restriction

  If Data Restriction is active, PR is denied.

- Malicious Call Trace

  If MCT is active, PR is denied.

- Recorded Announcement

  A call that is connected to a recorded announcement cannot have its path replaced.

- Trunk Access Code

  If the old connection was made using a TAC, PR is denied.

- Restriction Features

  PR is denied when restriction features such as COR of the Voice Terminal do not allow new connections to be established, unless the COS assigned to the old/new connections override the restrictions.

- Voice Terminals

  Voice terminal displays that show trunk group name should update with new trunk group information after PR occurs. Calling and connected party displays are not disturbed when PR takes place if the original display shows the connected party name, number, or both.

Transit Counter (TC)

- Call Forwarding (Diversion)

  When call forwarding (Diversion) occurs and the TC feature is enabled, the transit counter is set to zero.

- ISDN Trunk Group Administration

  If all of the conditions are satisfied for both the Tandem Hop Limitation and TC, TC takes precedence. In situations where the switch is an Incoming or Outgoing Gateway, either makes use of the hop count/transit count information provided by the other.

- Trunk Access Code (TAC)

  TC does not apply to TAC calls.

Call Completion (CC)

- Adjunct Switch Applications Interface (ASAI)

  ASAI cannot invoke/initiate QSIG-CC.

- Attendant Calling Waiting and Call Waiting Termination

  If you activate QSIG-CC to a single line voice terminal, the Attendant Call Waiting and Call Waiting Termination features are denied.

- Attendant Console Group

  You cannot activate QSIG-CC toward the attendant console group or towards an individual attendant extension.

- Attendant Control of Trunk Group Access

  You cannot activate QSIG-CC if the call uses a controlled trunk group.

- AUDIX

  You cannot activate QSIG-CC towards AUDIX. CC to any transferred-to station is not allowed.

- Automatic Call Distribution (ACD)

  You cannot activate QSIG-CC towards a voice terminal after dialing the ACD group extension. It is possible to invoke CC towards a station when dialing the individual's extension number. You can activate CC from any ACD agent.

- Bridged Call Appearance

  You cannot activate QSIG-CC from a bridged call appearance. When a call originates from a primary extension number, the return call notification rings at all bridged call appearances.

- Busy Verification

  After the called party in a QSIG-CC call hangs up, neither extension number 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).

- Diversion

  QSIG-CC requests are always activated at the principal user and not coverage points. Similar to ACB, QSIG-CC calls to the called user can redirect to coverage.

- Call Forwarding

  You cannot activate CCBS or CCNR towards a called station that has Call Forwarding enabled.

- Call Pickup

  On recall at the originating side, a group member cannot answer a QSIG-CC call for another group member.

- Call Waiting

  Call Waiting is denied when QSIG-CC is activated to the single-line voice terminal.

- Conference and Transfer

  You cannot activate QSIG-CC towards a transferred-to party.

- Hold

  A single-line voice terminal cannot receive a QSIG-CC call while it has a call on hold.

- Hotline Service

  A station originating a hotline service call cannot request CC.

- Internal Automatic Answer (IAA)

  If the IAA feature is enabled, QSIG-CC calls are not answered automatically.

- Manual Originating Line Service

  A manual originating service cannot request QSIG-CC.

- Multimedia Endpoints

  You cannot activate QSIG-CC towards multimedia data endpoints.

- Restriction Features
  - Class of Restriction (COR): Any terminal that is Origination-restricted cannot activate CC. Any terminal that is Termination-restricted cannot have CC activated towards it.
  - Class of Service (COS): To invoke CC, the **ACB** field on the **Class of Service** screen of the calling terminal must be set to *y*.
- Ringback Queuing

  Ringback Queueing and ACB share the same button to indicate that they are active. If the user has only one ACB button, then both features cannot be active at the same time.
- Outgoing Trunk Queuing

  Outgoing Trunk Queueing cannot be invoked after the calling party answers the priority call back call and no trunks are available. The CCBS and CCNR request cancels at both switches.
- Termination Extension Group (TEG)

  You cannot activate QSIG-CC towards a TEG extension, but QSIG-CC requests can be activated towards a single member in the group.
- Uniform Call Distribution and Direct Department Calling

  You cannot activate QSIG-CC towards a uniform call distribution group or a direct department calling group extension, but you can activate it when calling a single member in the group.
- Vector Directory Number (VDN)

  You cannot activate CC towards a VDN extension.

Message Waiting Indicators (MWI)

- Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) Partitioning

  All QSIG MWI messages use Partition Group 1 for routing.
- Alternate Facilities Restriction Levels

  QSIG MWI messages have unrestricted COR.
- DCP and Mode Code links to AUDIX

  QSIG MWI does not work with the DEFINITY AUDIX that emulates a DCP phone. A csi switch that communicates with an AUDIX system by using mode codes cannot be a QSIG message center switch complex.
- Authorization Codes

  The authorization codes do not block routing because the routing of temporary signaling connections (TSCs) used for QSIG MWI uses Facility Restriction Level 7 (FRL 7) (unrestricted).

- Automatic Alternate Routing (AAR)

  AAR may be used to route the QSIG TSCs.

- Automatic Route Selection (ARS)

  ARS may be used to route the QSIG TSCs.

- Call Coverage Features

  The served user switch uses call coverage paths to divert calls to users in the served user switch to the AUDIX hunt group on the Message Center switch.

- Class of Restriction (COR)

  QSIG MWI messages use the default COR of unrestricted.

- Class of Service (COS)

  QSIG MWI messages use the default COS of unrestricted.

- Facility Restriction Levels and Traveling Class Marks

  A QSIG MWI TSC always uses FRL 7 (unrestricted).

- Generalized Route Selection

  GRS uses the “TSC” column on the Route Pattern screen to select a preference for carrying QSIG MWI TSCs.

- ISDN - QSIG - BRI

  QSIG MWI is dependent on QSIG TSCs. QSIG MWI is possible over QSIG BRI lines.

- Message Sequence Tracer (MST)

  MST traces QSIG MWI messages.

- Off-Premises Station

  If a DS1 is used to implement an off-premises station, QSIG MWI does not work with the off-premises station. DS1 off-premise stations do not receive system message waiting indicators.

- Uniform Dial Plan (UDP)

  It is possible to route QSIG MWI messages by using UDP.

Called/Busy Name and Number

- Adjunct Switch Applications Interface (ASAI)

  A Connected Number is sent in the Connected Event to ASAI adjuncts. Therefore, upon receipt of a Called/Busy Number, it is stored in such a way that it is not sent accidentally as a Connected Number if no actual Connected Number is received in the CONNECT message when the call is answered.

- Call Diversion (including Reroute) for ISDN QSIG

  Both the Called Name and Called Number are sent to the ringing/busy extension.

- Call Transfer for ISDN QSIG

  Both the Called Name and Called Number of a ringing party is sent to the transferred-to party in the QSIG "Call Transfer Complete" message.

VALU Call Coverage

The interactions that apply to DCS call coverage apply to VALU call coverage, with the exceptions listed below. See Call Coverage interactions on page 251 in for more information.

- Call Coverage Off Premises

  Unlike DCS, QSIG-VALU can handle non-UDP numbers in the remote call coverage table. It is not limited to route only on UDP numbers.

- Consult

  Consult from the remote covering user to principal user is not supported.

- Displays

  When a Principal user bridges on the call, its display is updated with "CONFERENCE" and counted for the number of parties on the call. The remote covering user and calling user (local and remote) display is not updated with the word "CONFERENCE".

- QSIG Centralized Attendant Service (CAS)

  The calls that cover from a QSIG CAS branch to main are not treated as QSIG-VALU Coverage calls. This is because calls covered to "attd" (administered as a coverage point on a Coverage Path screen) do not use the Remote Call Coverage table and QSIG-VALU Call Coverage is supported only for coverage points associated with the Remote Call Coverage table. The implication of this is that the attendant on the main will lose QSIG-VALU Call Coverage display information and QSIG Path Replacement will not be invoked after the call is answered by the covering attendant.

- Coverage of Calls Redirected Off-Net (CCRON)

  If both QSIG-VALU coverage is enabled and CCRON is enable, the QSIG-VALU coverage will have a higher precedence than CCRON.

- Privacy - Manual Exclusion

  With the Call Coverage feature, when the principal user bridges onto a call that went to coverage and has been answered at the coverage point, the user is not dropped when Privacy - Manual Exclusion is activated by the Covering user.

  With QSIG-VALU Coverage, if the Principal bridges on the call after the remote covering user has answered the call. then the remote coverage user stays bridged until the call clears or the covering user goes on-hook.

- Simulated Bridged Appearance (SBA)

  With QSIG-VALU, maintaining SBA for Principal user will be based on the administration of the field **Maintain SBA at Principal** on the **System Parameters Call Coverage / Call Forwarding** screen.

  Note:

  SBAs are lost when Path Replacement occurs

- Temporary Bridged Appearance (TBA)

  Same interaction as Simulated Bridged Appearance.

- AUDIX / Centralized AUDIX

  The AUDIX system is usually specified as the last coverage point. When a call is routed to an AUDIX system (local or remote centralized place), the TBA is not maintained for the Principal user (that is, the Principal user can not bridge on to the call after it routes to the AUDIX system).

  For the last coverage point, which does not require control at the Principal user's switch, the QSIG-VALU Coverage shall divert the call as QSIG Diversion by Rerouting instead of QSIG Diversion by Forward-Switching and let the remote calling user's switch route the call directly to the remote covering number. If the Rerouting switch indicates failure, then the Principal user's switch (that is, the Served User's switch in terms of QSIG Diversion) shall revert to the normal QSIG-VALU Coverage handling. The advantage of this approach is that it saves trunk resources and provide path optimization without QSIG Path Replacement.

QSIG Centralized Attendant Service

- Abbreviated Dialing

  The main attendant can use abbreviated dialing buttons to extend QSIG-CAS calls.

- Attendant Auto-manual Splitting

  The attendant can split away from a call to call another party privately by pressing the START button.

- Attendant Auto Start and Don't Split

  The attendant can initiate a call while on an active call by pressing any button, without pressing the START button first. The system automatically splits the call and dials the next call. To deactivate Auto Start, press the **Don't Split** button.

- Attendant Backup Alerting

  If attendant backup alerting is turned on, other users on the main may have the ability to answer attendant seeking calls.

- Attendant Call Waiting

  Attendant call waiting is available for calls that originate on the main.

- Attendant Calling of Inward and Public Restricted Stations

  A user who is inward restricted cannot receive a call originated or extended by the attendant at the QSIG CAS main. A user who is public restricted is able to receive calls originated and extended by the QSIG CAS main attendant, provided these calls are routed over QSIG ISDN tie trunks.

- Attendant Conference

  By using the attendant split/swap feature, it is possible for the attendant to conference join the attendant, calling party, and extended party together in conference. If the attendant drops out of the conference, leaving just the calling party and extended party, path replacement is not attempted.

- Attendant Direct Extension Selection (DXS) With Busy Lamp (standard and enhanced)

  For QSIG-CAS the DXS allows attendants to monitor and place calls to users on the main and to place calls to users on a branch only when UDP is used.

- Attendant Group and Tenant Partitioning

  Attendant Group and Tenant Partitioning are local features that do not require QSIG signaling.

  Attendant Group and Tenant Partitioning do not function on a CAS branch. You can administer tenant partitioning and multiple attendant groups on a branch. However, all attendant-seeking calls at the branch are directed to the QSIG-CAS number, as administered on the **console-parameters** screen, regardless of any tenant partition. If the QSIG-CAS number corresponds to the Dial Access to Attendant number at the main or to a

Vector Directory Number (VDN) that eventually routes to the Dial Access to Attendant number at the main, the call is directed to the attendant group assigned to the tenant partition of the incoming trunk to the main.

- Attendant Interposition Calling and Transfer

  Attendant Interposition calling and transfer is a local feature that remains unchanged by QSIG-CAS. Attendants on the main still have the ability to call and transfer to each other using Individual Attendant Extensions.

- Attendant Intrusion

  Intrusion is not available in QSIG-CAS to calls that are incoming from a branch.

- Attendant Misoperation

  Misoperation is used only in France and Italy. It is a local feature and does not require QSIG signaling. If the system goes into Night Service while an attendant has a call on hold, the call re-alerts at the attendant console. If it is unanswered after an administrable amount of time, the call begins alerting at the night service destination.

- Attendant Override of Diversion

  Override of Diversion is not available in QSIG-CAS for incoming calls from a branch.

- Attendant Recall

  Attendant Recall is not available in QSIG-CAS to calls incoming from the branch.

- Attendant Release Loop Operation

  Attendant Release Loop Operation is a local switch feature. It allows an unanswered extended call on the main to return to the attendant after an administrable amount of time. The call first tries to return to the same attendant that originally answered the call and, if that attendant is not available, the call goes to the next available attendant (waiting in the Attendant Queue if necessary).

- Attendant Return Call

  Attendant Return Call functions in the following manner: Suppose a call comes into the attendant from a branch. If the attendant extends the call and it is unanswered after an administrable amount of time the call returns to the attendant. Initially, the call attempts to return to the same attendant that originally handled the call. If that attendant is unavailable, then the call goes to the next available attendant (waiting in the Attendant Queue if necessary).

- Attendant Serial Calling

  Attendant Serial Calling is not available in QSIG-CAS to incoming calls from the branch.

- Attendant Tones

  Call identification tones are not heard by attendants answering calls from a QSIG-CAS branch.

- Attendant Trunk Group Busy/Warning Indicators

  The attendant can only receive busy/warning indicators for trunks at the main. The attendant cannot receive information about branch trunks.

- Attendant Vectoring

  The attendant vectoring feature is available to QSIG-CAS at the branch and the main. An attendant-seeking call terminating at the main follows any vector steps that are defined at the main.

  The QSIG-CAS Number should not contain the number of a remote VDN. Note that there is no check to block such administration, but QSIG CAS may not function correctly.

- Automatic Circuit Assurance (ACA)

  The CAS attendant cannot receive ACA referral calls from a branch because any administered ACA referral extension must be local.

- Call coverage

  The attendant group is allowed to be a coverage point.

  If the call diverts from the branch to the main over a non-QSIG ISDN trunk, then the call is treated as a forwarded call. That is, Call Coverage Off Net (CCRON) procedures do not apply and the call is not brought back to the branch.

  Note:

  > In order to obtain the full functionality of QSIG CAS, it is recommended that routing patterns are set up so that a QSIG trunk is used when sending a call from the branch to the main.

  If the call diverts from the branch to the main over a QSIG trunk (not QSIG VALU), then QSIG Diversion procedures apply.

  If the call diverts from the branch to the main over a QSIG VALU trunk, then QSIG VALU Call Coverage procedures apply.

- Call forwarding

  Forwarding calls to the QSIG-CAS number is allowed.

- Call park

  If a call is parked and the Call Park Timeout Interval (as set on the **Feature Related System Parameters** screen) expires, the call is sent to the attendant.

- Call Record Handling Option

  Calls are sent to the attendant as non-CDR calls if the following conditions all hold:

  - the call is subject to CDR, **and**
  - the CDR buffer is full, **and**
  - the attendant is administered as the **Call Record Handling Option** on the **CDR System Parameters** screen.

- CDR Reports

  The format of the CDR data report is an administrable option on the CDR Systems Parameters screen. Customers can select from a list of pre-defined formats or create their own. The content of the CDR records is unchanged by QSIG-CAS.

  CDR records generated at the main are covered by existing procedures. Calls incoming to the attendant look like incoming trunk calls. Calls originated or extended by the attendant look like outgoing calls.

- CAS Back-Up Extension

  The CAS Back-Up Extension is used in a Release Link Trunk (RLT)-CAS environment but has no benefit in QSIG-CAS.

- Conference

  If a user on a branch conferences an attendant onto a call, the attendant’s display is not updated with “conference”. There is no QSIG standard defined for Conference and Avaya has not implemented conference using Manufacturer Specific Information (MSI).

- Centralized AUDIX

  When a user zero’s out of AUDIX, if the destination is the attendant and the host is a QSIG-CAS branch, then the call is sent to the QSIG-CAS attendant.

- DCS+

  On an incoming attendant-seeking call, calling-party information may be received at the branch if a call comes over a DCS+ trunk in the network.

- Dial Access to Attendant

  When a user on a branch dials the Dial Access to Attendant number, as administered on the **Dial Plan Analysis** screen, the call is sent to an attendant on the main.

- DID/Tie/ISDN Intercept

  DID, Tie, and ISDN trunk calls that are intercepted are sent to the attendant on the main.

- Emergency access to attendant

  Emergency access may be administered so that if stations are off hook for an extended period of time, then a call is placed to the attendant, or a user can dial an Emergency access to attendant feature access code. Emergency access to calls invoked at a CAS branch attendant do not go to the attendant on the main. Instead, the call goes to an attendant on the branch. If there is no branch attendant, the call is denied.

- Individual Attendant Access

  An attendant may be assigned an individual extension so that it is possible to dial that attendant directly rather than dialing the attendant group.

- ISDN (non-QSIG)

  On an incoming attendant-seeking call, calling party information may be received at the branch for a call coming in over an ISDN trunk.

- Leave word calling (LWC)

  System-wide LWC Message Retrieval is not available at the CAS main attendant for a branch user’s messages.

- Malicious Call Trace (MCT)

  MCT is a feature that works on existing calls. MCT will work in QSIG-CAS provided the attendants performing MCT-Activate, MCT-Control, and MCT-Deactivate are all on the same switch. That is, an attendant on the main cannot work with an attendant on the branch to perform MCT. ETSI MCT and Australia MCT cannot be invoked remotely either.

- Multifrequency Signaling

  Calls coming into a branch over Multifrequency trunks are subject to intercept and may be sent to the attendant at the main. Multifrequency signaling can indicate that an incoming call on a multifrequency (MF) trunk terminate at the attendant, regardless of the dialed extension.

- Night Service

  Night Service is available to QSIG-CAS. If a branch is in night service, then all attendant-seeking calls for that branch are routed to the night service destination, not the CAS attendant. If the main is in night service, then all attendant seeking calls at the main (either incoming from the main or branch) are routed to the night service destination. The night service destination must be local.

  Communication Manager supports the following night service features:

  - Hunt Group Night Service — allows an attendant to assign a hunt group to night service
  - Night Console Service — allows a console to be designated as the night service destination
  - Night Station Service — allows a station to be designated as the night service destination
  - Trunk Answer from Any Station (TAAS) — allows voice terminal users to answer attendant seeking calls
  - Trunk Group Night Service — allows an attendant or designated night service terminal user to assign one or more trunk groups to night service

- Outgoing Trunk Queuing

  Attendant-seeking calls from branch to main can be queued at the outgoing branch trunk group.

- QSIG

  All the existing QSIG features and services are available in QSIG-CAS. QSIG-CAS is available in any QSIG-CAS ISDN network (PRI, BRI, PRI/ATM, and IP).

- QSIG Call Offer

  Calls extended by the attendant can invoke Call Offer. If a call invokes Call Offer, attendant return call procedures still apply.

- Extending a Call

  QSIG CAS ensures that QSIG Path Replacement is attempted after split/swap, provided that all three parties (original calling party, the attendant, and the called party) are never conferenced together. That is, if the attendant toggles between the other two parties for any number of times, never conferencing all three together, and then joins the two parties together (with the attendant now out of the picture and ready to handle other calls), Path Replacement is attempted.

- Security Violation Notification (SVN)

  The CAS attendant cannot receive SVN referral calls from a branch. Any administered SVN referral extension must be local.

- Special Application 8140 - Attendant Dial 0 Redirect

  Attendant Dial 0 Redirect allows calls to the attendant group to be routed to one of two attendant groups based on their call priority level, and to alert with emergency ring. The two groups are the default attendant group and the priority attendant group. Administration of whether a priority level routes to the priority group is done on the **Console Parameters** screen.

  Administration on the **Console Parameters** screen at the main determines to which attendant group the priority level routes, and whether calls of that priority level alert with emergency tone.

- Special Application 8141 - Listed Directory Number (LDN) Attendant Queue Priority

  Calls coming to the main from a QSIG-CAS branch cannot be queued by LDN Priority. QSIG-CAS does not change the ability to of LDN Queue Priority to function for calls coming directly into the main.

- Special Application 8156 - Attendant Queuing by COR

  Calls coming to the main from a QSIG-CAS branch cannot be queued by COR Priority. QSIG-CAS does not change the ability to of Attendant Queueing by COR to function for calls coming directly into or originating at the main.

- Timed reminder and Attendant timers

  Attendant timers are:

  - Timed Reminder on Hold — starts when an attendant puts a call on hold. When this timer expires, the held call alerts the attendant.
  - Return Call Timeout — starts when a call is extended and then released from an attendant console. If this timer expires, the call is returned to the attendant.
  - Time In Queue Warning — indicates the amount of time a call can wait in the attendant queue before activating an alert.

  - No Answer Timeout — Calls that terminate at an attendant console ring with primary alerting until this timeout value is reached. When this timeout value is reached, the call rings with a secondary, higher pitch.
  - Alerting — notifies, using secondary alerting, other attendants in an attendant group of an unanswered call. The Attendant Alerting Timed Reminder starts when a call reaches the Attendant No Answer Timeout maximum value.
- Transfer Out of AUDIX by Dialing 0

  Attendant seeking calls that transfer out of AUDIX by dialing 0, whose host switch is a branch, are sent to the QSIG-CAS attendant on the main whenever the dial 0 out of AUDIX destination corresponds to the attendant group.

Centralized Attendant Service (CAS) interactions and considerations

CAS Interactions

- Abbreviated Dialing

  The main attendant can use an Abbreviated Dialing button to extend CAS calls after obtaining branch dial tone.
- Attendant Auto-Manual Splitting

  The SPLIT lamp and button do not function on CAS main calls extended using the RLT trunk. 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, new attendant-seeking calls route 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 and Automatic Route Selection

  CAS calls can be routed using AAR and ARS.

- 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

  Redirect calls to a centralized attendant by Call Coverage. Do not redirect calls to a CAS backup extension for backup service using Send All Calls to the backup extension’s coverage path.

- Call Detail Recording (CDR)

  If the CAS main RLT trunk has the CDR option selected, CDR records generate for incoming CAS calls.

- Call Forwarding

  Do not forward calls to a CAS extension.

- DCS Operation

  If an RLT trunk group is administered as a DCS trunk, the following interaction applies: On an incoming CAS call to the attendant, the DCS message displays instead of the name of the incoming RLT trunk group. Upon answering the call, the attendant hears call-identification tones, indicating that the call is a CAS call. Use a **TRUNK-NAME** button to obtain the name of the RLT trunk group.

- Direct Extension Selection (DXS) and Direct Trunk Group Selection (DTGS) Buttons

  DXS and DTGS buttons at the main attendant console can be used with CAS. However, with DXS buttons, it takes a few seconds before the attendant hears ringback tone.

- Emergency Access to the Attendant

  CAS Branch Emergency Access calls generated by a Feature Access Code route Off-Hook Alert to the branch attendant group. If there is no attendant in the branch, the call routes to the branch’s administered Emergency Access Redirection Extension. When the branch switch is in CAS Backup Service, the calls route to the backup station and the call is treated as a normal call.

- Hunt Groups

  If an incoming CAS call directs to a hunt group, the call does not redirect to the hunt group’s coverage path. Depending on the circumstances, the attendant can get a busy tone or ringing.

- Leave Word Calling

  If a message is left for a branch user and the attendant at the CAS switch tries to retrieve the message by using LWC message retrieval, permission is denied.

- Night Service — Night Console Service

  When the CAS main enters night service, CAS calls terminate at the CAS main night-service destination. When the branch enters Night Service, CAS calls route to the branch night console, the LDN night station, or the Trunk Answer from Any Station (TAAS).

- Night Service — Trunk Answer from Any Station (TAAS)

  In a multiswitch DCS environment with CAS, the result of transferring incoming trunk calls using Night Service Extension or Trunk Answer from Any Station varies depending on the home switch of the transferred-to station, the home switch of the connected trunk, and the type of night-service function chosen (Night Service Extension, Trunk Answer from Any Station, or both).

- Non-attendant Console Handling of CAS Calls

  The CAS branch calls terminate at the CAS main based on the incoming RLT trunk-group day destination or night-service destination. You can also answer a CAS call by the Trunk Answer from Any Station feature.

CAS considerations

Branch Attendants

- 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 but can be routed back to them by the centralized attendant.
- Branch calls terminate on the CAS main switch based on the incoming RLT trunk-group day-destination or night-service destination. An attendant console is not always answering or extending incoming CAS calls. If someone other than an attendant answers a CAS call, that person can extend the call back to the branch by pressing the FLASH button on a multi-appearance voice terminal or flashing the switchhook on a single-line voice terminal. 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.
- When an analog-station call goes to coverage, the station drops from the call. This is the exception to the branch leaving the extended-to party ringing. If the main attendant extends a call to an analog station and that call goes to coverage and later returns to the main attendant, the call is treated as an incoming LDN call and the attendant must re-extend the call, if requested by the user.
- On an incoming CAS call to the main attendant, the **Name** field from the **Trunk Group** screen for that RLT displays to the attendant. Therefore, you should administer the field to provide meaningful branch identification information.
- Music-on-Hold feature at branch applies to two stages of LDN calls: during call extension and Remote Hold.

Italian TGU/TGE (main and satellite) interactions

Table 34: Italian TGU/TGE feature interactions and considerations on page 272 contains important feature interactions and considerations for Italian TGU/TGE (main and satellite).

Table 34: Italian TGU/TGE feature interactions and considerations

Feature	Description	Interactions	Considerations
Attendant Call Waiting		Call Waiting is provided through Italian TGU/TGE (main and satellite) trunks. Call Waiting also is provided in Italy and all other countries through DCS.	
Attendant Intrusion		Attendant Intrusion is provided on satellite switches using TGU/TGE trunks. Attendant Intrusion also is provided through DCS.	

Hairpinning and shuffling feature interactions

Any 50 to 200 ms break in the speech path resulting from shuffling or hairpinning also affects certain features. In addition, while a shuffled or hairpinned call is in progress, certain user actions might require the switch to redirect the call to the TDM bus. Table 35: IP hairpinning/ shuffling feature interactions on page 273 describes several feature interaction scenarios.

Table 35: IP hairpinning/shuffling feature interactions ***1 of 25***

	Conditions		**Result**
Feature	**If**	**And(s)**	**Then**
Abbreviated Dial	A call has been established	An end user presses an abbreviated dialing button	The digits stored in the abbreviated dial button are inserted into the call. Refer to End-to-End Signaling on page 286 for other interactions.
Attendant Console - Attendant busy lamp field - Busy Indication button	A station is on a shuffled or hairpinned connection		The **`status station`** ***`<extension>`*** reports shows the **Attendant Busy Lamp** field as `busy`, and the **Busy Indication** buttons are on.
- Attendant Intrusion Scenario A	Endpoints A and B have a shuffled or hairpinned connection	An attendant attempts to use intrusion to break into the conversation and There are no audio channels available on a TN2302AP in A’s network region, or there are no audio channels available on a TN2302AP in B’s network region	The switch gives the same reorder tone and lamp flashes to the attendant as the switch would do if A and B had been circuit switched and a similar attempt failed for lack of switch resources.

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Table 35: IP hairpinning/shuffling feature interactions ***2 of 25***

	Conditions		**Result**
Feature	**If**	**And(s)**	**Then**
● Attendant Intrusion Scenario B	Endpoints A and B have a shuffled or hairpinned connection	An attendant attempts to use intrusion to break into the conversation, and There is 1 audio channel available on a TN2302AP IP Media Processor circuit pack(s) in each of A's and B's network regions,	The switch redirects the audio stream of both endpoints A and B back to a TN2302AP Prowler port each, and connects the attendant into the call.
● Attendant Recall	N/A	N/A	Attendant Recall only applies to calls held at an attendant console, which do not shuffle or hairpin audio connections.
Automatic Callback	Endpoints A and B have a shuffled or hairpinned connection	Endpoint A is a single line set, and A third party, C, attempts to reach endpoint A, and C activates automatic callback, and A eventually becomes idle, causing endpoint C to ring, and C eventually answers the automatic callback call, and There are no audio channels available on a TN2302AP circuit pack in A's network region to allow a call to A when C answers the automatic callback call	C receives the same reorder tone and lamp flashes as the switch would provide if A and B had been circuit switched and a similar attempt failed for lack of switch resources. C is able to restart automatic callback toward A.
			2 of 25

Table 35: IP hairpinning/shuffling feature interactions ***3 of 25***

	Conditions		**Result**
Feature	**If**	**And(s)**	**Then**
Automatic Call Distribution (ACD) ● Assist	If endpoint A and endpoint B have an audio connection to each other through an ip-prowler-ip hairpin or ip-ip directly	User A presses the assist button	B is placed on soft hold awaiting a conference, and the switch launches a call from A toward the split supervisor. See Conference for interactions between shuffling, hairpinning, and conference.
● Multiple Call Handling			Interactions between shuffling, hairpinning, and ACD multiple call handling are the same as between these features and Hold.
Bridging ● Circuit-switched endpoint bridged with IP endpoint Scenario A	Endpoints A and B have a shuffled or hairpinned connection	User C has a bridged call appearance of endpoint A on C’s set	A shuffled or hairpinned connection between A and B is possible. User C is not considered a 3rd party to this call unless user C selects the bridged call appearance.

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Table 35: IP hairpinning/shuffling feature interactions ***4 of 25***

	Conditions		**Result**
Feature	**If**	**And(s)**	**Then**
• Circuit-switched end-point bridged with IP end-point Scenario B	Endpoints A and B have a shuffled or hairpinned connection	Circuit-switched (TDM) user C has a bridged call appearance of set A on C's set, and C presses that call appearance button	• If there is 1 audio channel available for A on a TN2302AP circuit pack in A's network region, and there is 1 audio channel available for B on a TN2302AP in B's network region, then the switch redirects the audio streams of both endpoints A and B back to a TN2302AP port each. All three parties are able to hear each other. • If there are 0 audio channels available on a TN2302AP circuit pack in A's network region, or there are 0 audio channels available on a TN2302AP in B's network region, then set C's bridged call appearance lamp flashes, and the two endpoints A and B remain directly ip-ip connected or connected together through an ip-prowler-ip hairpin connection. C hears switch generated reorder tone. The audio path between endpoints A and B remains connected until the path(s) back to the TN2302AP(s) are allocated. The same kind of interaction occurs if IP endpoints C and B are initially connected together using circuit switched endpoint A's bridged call appearance, and A attempts to select that appearance.

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Table 35: IP hairpinning/shuffling feature interactions ***5 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
● Bridging--All endpoints are IP	Endpoints A and B have a shuffled or hairpinned connection	User C has a bridged call appearance of set A on C's set, and C presses that call appearance button,	● If there is 1 audio channel available for C on a TN2302AP in C's network region, the switch proceeds as described in Scenario A (bridging circuit-switched endpoints with IP endpoints above). ● If there are no audio channels available on a TN2302AP in C's network region, then set C's bridged call appearance lamp flashes, and A and B remain shuffled or hairpinned. Depending on the type of endpoint that C is using, C might hear a locally-generated tone such as reorder tone or Microsoft Windows' "program error" sound.
● Bridging--New connections			The switch cannot set up a shuffled or hairpinned connection between two endpoints while either endpoint is bridged in a call with additional parties. This scenario constitutes a 3-party conference call.
Busy Verification	Endpoints A and B have a shuffled or hairpinned connection	Endpoint A is a single line set, and A third party, C, attempts to use busy verification to reach endpoint A, and There is 1 audio channel available for A on a TN2302AP in A's network region, and there is 1 audio channel available for B on a TN2302AP in B's network region,	The switch redirects the audio streams of both endpoints A and B back to a TN2302AP Prowler port each. The audio path between endpoints A and B remains connected until the path(s) back to the TN2302AP(s) are allocated. Once the audio paths of both A and B are back on the switch, the third party C is bridged into the call.

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Table 35: IP hairpinning/shuffling feature interactions ***6 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Call Coverage	An incoming call to a principal user is redirected by call coverage	The covered-to party answers	Call can be shuffled if the Determining if shuffling is possible on page 86 are met. If the principal user has a simulated bridge appearance activated, then the interactions in Bridging are applicable.
Call Forwarding	An incoming call to a user is redirected by call forwarding	The forwarded-to party answers,	Call can be shuffled if the Determining if shuffling is possible on page 86 are met.
			6 of 25

Table 35: IP hairpinning/shuffling feature interactions ***7 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Call Park ● Parking action	Endpoints A and B have a shuffled or hairpinned connection	User A presses the call park button, and there is no call parked on A’s station previously,	● If there is 1 audio channel available for A on a Prowler board in A’s network region, and there is 1 audio channel available for B on a TN2302 in B’s network region, the switch will redirects the audio streams of both A and B back to a TN2302 port each. A and B both hear confirmation tone and maintain a two-way talk path. If A hangs up, B remains in the call-park state, and B remains connected to the TDM bus. ● If there are no audio channels available on a TN2302 in B’s network region or there are no audio channels available on a TN2302 in the A’s network region, then that endpoint(s) will not hear confirmation tone, but set A's call park lamp turns on. A and B remain shuffled or hairpinned. If A hangs up, B remains in the call-park state. If instead of using a call park button, A puts B on hold or uses conference or transfer to put B on soft hold, and then successfully gets dial tone from the switch and dials the call park FAC, interactions described in Hold apply.
Call Park ● By Attendant	Endpoints A and B have a shuffled or hairpinned connection	An attendant parks a call from C at endpoint A's extension	The Call Park button lamp (if provided on set A) is on. As long as user A ignores the parked call, endpoints A and B remain shuffled or hairpinned.

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Table 35: IP hairpinning/shuffling feature interactions ***8 of 25***

	Conditions		**Result**
Feature	**If**	**And(s)**	**Then**
Call Park ● Un-parking	If user A attempts to access a parked user C,		● If user A has access to dial tone and attempts to reach C by dialing the answer-back FAC, then this is the same as if user A had attempted to reach another user by dialing that party's extension number. ● If user A has a call parked, and if there are no audio channels available on a TN2302AP in A's network region when A selects the call park button, then set A's call park lamp flashes. Depending on A's equipment type, A may hear a locally-generated tone such as reorder tone or Microsoft Windows' "program error" sound. If A and B are talking to each other after B is parked, and a third party C goes off hook and dials the call-park-answer-back feature access code and A's extension, then the result is a 3-way conference call.
Call Pickup and Directed Call Pickup	User A attempts to call user B	User C uses call pickup to answer the call, and The **Temporary Bridged Appearance on Call pickup** field on the **System Parameters Features** form is *y* (yes)	Bridged appearance on set B of the call between user A and user C is maintained (see Bridging).
			8 of 25

Table 35: IP hairpinning/shuffling feature interactions ***9 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Call Vectoring - Return Destination	Endpoint B calls into the switch as an incoming H.323 trunk call	Endpoint B's call is handled by call vectoring, and Endpoint B ends up hairpinned with endpoint A, and The VDN had an administered Return Destination, and Endpoint A hangs up, and There is 1 audio channel available on a TN2302AP in B's network region	B's audio stream is redirected back to a TN2302AP port, and the call is connected to the administered return destination. If there are no audio channels available on a TN2302AP in B's network region when A hangs up, then B receives the same reorder tone or coverage treatment as if A had been circuit-switched and a similar attempt failed for lack of switch resources.
Call Vectoring - VDN of Origin Announcement (VOA) Scenario A	Endpoints A and B have a shuffled or hairpinned connection	User A presses the VOA Repeat key, and There are no audio channels available on a TN2302AP in either A's or B's network region	The switch flutters the voa-repeat lamp.
Call Vectoring - VDN of Origin Announcement Scenario B	Endpoints A and B have a shuffled or hairpinned connection	User A presses the VOA Repeat button, and There is 1 audio channel available for A on a TN2302AP in A's network region, and there is 1 audio channel available for B on a TN2302AP in B's network region,	Both A's and B's audio streams are redirected back to a TN2302AP port each, user A is connected to the VOA announcement if this redirection can be completed in 7 seconds (the maximum time currently allowed for VOA over ATM). If the redirection takes longer than 7 seconds, then the announcement plays anyway, and the voa-repeat lamp flutters. When the announcement completes, the audio connection is shuffled or hairpinned.

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Table 35: IP hairpinning/shuffling feature interactions ***10 of 25***

	Conditions		Result
Feature	If	And(s)	Then
Call Waiting - Scenario A	Endpoints A and B have a shuffled or hairpinned connection	Endpoint A is a single line set administered for call waiting, and A third party, C, attempts to reach endpoint A, and There are no audio channels available on a TN2302AP in A's network region, or there are no audio channels available on a TN2302AP in B's network region.	Third party (C) receives the same ringback tone as if A and B had been circuit-switched and a similar attempt failed for lack of switch resources. User A does know that there was an attempt to call him unless the third party uses some other type of notification method, such as leave word calling.
Call Waiting - Scenario B	Endpoints A and B have a shuffled or hairpinned connection	Endpoint A is a single line set administered for call waiting, and A third party, C, attempts to reach endpoint A, and There is 1 audio channel available for A on a TN2302AP in A's network region, and there is 1 audio channel available for B on a TN2302AP in B's network region.	Audio streams of both endpoints A and B are redirected back to a TN2302AP port each. The audio path between endpoints A and B remains connected until the path(s) back to the TN2302APs are allocated. Only after endpoint A's audio stream is connected back to the TN2302AP will the system play call waiting tone to endpoint A. A and B are able to continue to talk to each other while the tone is playing.
Code Calling (Chime Paging)			Calls parked through code calling have the same interaction with hairpinning and shuffling as do calls parked through the call park feature (see Call Park)
			10 of 25

Table 35: IP hairpinning/shuffling feature interactions ***11 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Conference	Endpoints A and B have a shuffled or hairpinned connection	User A presses the conference button **NOTE:** The switch cannot set up a shuffled or hairpinned connection between 2 endpoints while either is conferenced with additional parties.	● If there is 1 audio channel available for - A on a TN2302AP in A's network region, the audio stream of endpoint A is redirected back to a TN2302AP port. A new call appearance lamp turns on immediately on set A, and A hears dial tone on the new call appearance. - B on a TN2302AP in B's network region, and if B is administered for music on hold, the audio stream of endpoint A is redirected back to a TN2302AP port. ● If there are no audio channels available for - B on a TN2302AP in B's network region, and if B is administered for music on hold, the switch gives B silence on hold. - A on a TN2302AP in A's network region, then a new call appearance on set A flashes. Depending on the type of set that A is using, A may hear a locally-generated tone (such as reorder tone or "program error" sound).

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Table 35: IP hairpinning/shuffling feature interactions ***12 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
			At this point user A may choose to ● Reselect the held call appearance and be reconnected to B, or ● Wait a few seconds and then try another call appearance should a TN2302AP audio channel have since become available. The audio path between endpoints A and B remains connected until the path(s) back to the TN2302AP are assigned. Only IP Softphone with integrated audio, IP telephone, telecommuter and Road Warrior endpoints can conference calls; simple H.323 stations cannot use conference.
Conference ● Soft Hold			The switch does not set up a shuffled or hairpinned connection between two endpoints while either endpoint has a TDM party on soft hold awaiting a conference. However, the switch can set up a shuffled or hairpinned connection between two endpoints if either endpoint only has IP parties on soft hold awaiting a conference. For example, both IP endpoints A and B are shuffled. If A presses the conference button, gets dial tone, and calls C, then the call from A to C can transition to a shuffled connection. If A now presses the conference button a second time, but now there are no TN2302AP ports available, the conference attempt fails. There is no lamp to flutter on a conference button, but the line appearance lamp does flutter. A and C are now talking to each other, and B is still on soft hold.

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Table 35: IP hairpinning/shuffling feature interactions ***13 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Conference ● Attendant Conference			A 2-party call held on the attendant console cannot be shuffled or hairpinned (see Hold).
Conference ● Conference on Hold	If IP endpoint A has only IP endpoint B on hold	B is not using a TN2302AP port, and User A presses the conference button,	● If there is 1 audio channel available for A on a TN2302AP in A's network region, the audio stream of endpoint A is redirected back to a TN2302AP port. A new call appearance lamp lights immediately on endpoint A, and A hears dial tone on the new call appearance. ● If there are no audio channels available for A on a TN2302AP in A's network region, then a new call appearance on set A flashes. Depending on the type of set that A is using, A may hear a locally-generated tone (such as reorder tone or a "program error" sound). At this point user A may choose to - Reselect the held call appearance and reconnect to B, or - Wait a few seconds and then try another call appearance should a TN2302AP audio channel have since become available.
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Table 35: IP hairpinning/shuffling feature interactions ***14 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Consult	Endpoints A and B have a shuffled or hairpinned connection	User A presses the consult button	If there are no audio channels available on a TN2302AP in A's network region, then both endpoint A's consult button lamp and the call appearance that was attempted for the call to C flash. Depending on the type of set that A is using, A may hear a locally-generated tone (such as reorder tone or a "program error" sound)
Distributed Communications System (DCS)			● DCS Automatic Callback: see Automatic Callback. ● DCS Busy Verification: see Busy Verification. ● DCS Call Waiting: see Call Waiting. ● DCS Multi-appearance Conference: see Conference. ● DCS Multi-appearance Transfer: see Transfer. ● Italian DCS Attendant Intrusion: Attendant Console.
End-to-End Signaling	Endpoints A and B have a shuffled or hairpinned connection	User A presses a phone keypad (DTMF) button	The switch ensures that the DTMF signal or its equivalent reaches the far end(s) of the connection. In an IP-TDM call, the TN2302AP only detects DTMF tones from the TDM bus, not from the IP side.
Fax			For endpoints known to be used for fax, the safest thing to do is to administer those endpoints to prevent shuffling during the potential 200ms break in the middle of the call.

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Table 35: IP hairpinning/shuffling feature interactions ***15 of 25***

	Conditions		**Result**
Feature	**If**	**And(s)**	**Then**
Hold ● Scenario A	Endpoints A and B have a shuffled or hairpinned connection	User A presses the hold button, and B is administered for music on hold, and There is 1 audio channel available on a TN2302AP in B's network region	B's audio stream is redirected back to the TN2302AP port. If there are no audio channels available on a TN2302AP in B's network region, then the switch puts B on hold, but B hears silence.
Hold ● Scenario B	If user A, user B, and user C are talking on a conference call	User C presses the hold button, leaving A and B talking together	A shuffled or hairpinned call cannot be set up between those two endpoints as long as C keeps them on hold. This prevents a delay when C re-enters the call. **NOTE:** Any form of hold that involves TDM endpoints blocks a transition to shuffled or hairpinned connections. For example, if IP set A holds TDM set B, then a call from A to IP set C remains IP-TDM. But if IP set A holds IP set B, then a call from A to IP set C can be shuffled or hairpinned. Similarly, if IP sets A, B, and C are conferenced on the TDM bus, and A puts the B-C call on hold, the B-C call remains on the TDM bus.

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Table 35: IP hairpinning/shuffling feature interactions ***16 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Hold ● Automatic Hold	Endpoints A and B have a shuffled or hairpinned connection	IP endpoint C calls endpoint A, and A and C are both eligible for a shuffled or hairpinned connections between them, and The switch is administered for automatic hold,	● If there is 1 audio channel available for A on a TN2302AP in A's network region, the switch rings endpoint A. If A selects the ringing call appearance, the audio stream of endpoint A is redirected back to the TN2302AP port. User A and user C are connected through IP, leaving user B on hold. ● If there are no audio channels available for A on a TN2302AP in A's network region, the switch supplies the same reorder tone or coverage treatment to C as the switch would provide if A had been circuit switched and a similar attempt failed for lack of switch resources.
Intercom			An intercom call between two endpoints is shuffled or hairpinned in exactly the same way as a regular 2-party call.
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Table 35: IP hairpinning/shuffling feature interactions ***17 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Malicious Call Trace (MCT) ● Scenario A	Endpoints A and B have a shuffled or hairpinned connection	User B gets malicious, so someone (either user A or a third party) initiates MCT for user A's extension, and There is a voice recorder port available on the switch, and There are no audio channels available on a TN2302AP in either A's or B's network region	MCT proceeds but the switch does not attempt to connect a voice recorder into the conversation.
Malicious Call Trace (MCT) ● Scenario B	Endpoints A and B have a shuffled or hairpinned connection	User B gets malicious, so someone (either user A or a third party) initiates MCT for user A's extension, and There is a voice recorder port available on the switch, and There is 1 audio channel available for A on a TN2302AP in A's network region, and there is 1 audio channel available for B on a TN2302AP in B's network region,	A's and B's voice path is redirected back to the switch in order to bridge a voice recorder into the call. Doing so on a shuffled or hairpinned call puts an approximately 200 ms break into the speech, risking the malicious caller noticing the break in conversation. If an IP endpoint has been receiving malicious calls, the system administrator might want to administer that endpoint to prevent shuffling. If the endpoints are hairpinned instead of shuffled, then the break in conversation should be short enough not to be noticed. If an H.323 trunk is involved in the MCT, this resource is blocked from being dropped from the switch side to facilitate the tracing activity.
Manual Signaling	Endpoints A and B have a shuffled or hairpinned connection	A third endpoint C uses manual signaling to ring endpoint A,	A and B remain shuffled or hairpinned.
Multimedia Call Handling (MMCH)			A MMCH call requires access to the TDM bus, so shuffled or hairpinned connections are not possible.

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Table 35: IP hairpinning/shuffling feature interactions ***18 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Music On Hold and Tenant Partitioning			Music on Hold quality deteriorates through some codecs, particularly G.723. If a customer wants to provide news or silence on hold for endpoints using G.723, but music on hold for endpoints using any other codec, they can partially do so through Tenant Partitioning: 1. Administer the switch to have two tenant partitions. 2. Place endpoints whose codec list *does not* include G.723 into one tenant partition, and place those endpoints whose codec *includes* G.723 into another tenant partition. 3. Administer the partitions so that both have full permission to call the other. 4. Administer different music sources for the two tenant partitions, and ensure that the tenant partition that allows G.723 only has suitable-sounding audio material.
Outgoing Trunk Queuing	A station user uses outgoing trunk queuing toward an IP trunk		The switch waits for a signaling channel (a B channel) to become available. If, when a signaling channel becomes available, there are no voice channels (medpro or TN2302AP Prowler channels) available, the station user will receive the same reorder tone and lamp flashes as the switch would provide if the trunk group had been circuit switched and a similar attempt failed for lack of switch resources.
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Table 35: IP hairpinning/shuffling feature interactions ***19 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
QSIG	Two Avaya servers are interconnected by a trunk that supports QSIG.		The QSIG APDUs are transported across that interface. QSIG APDUs are transmitted regardless of whether the trunk is TDM-connected or IP-IP-connected.
QSIG ● Path Replacement			QSIG Path Replacement offers certain advantages over shuffling from a TDM connection. After shuffling, the signaling path is not changed and the resources such as H.323 trunk are not released. In scenarios such as call transfer, call forwarding, call coverage, call transiting through other Avaya equipment, if QSIG Path Replacement can be invoked, it may provide a direct media and signaling connection. For example, if a call exists from switch A to switch B to switch C, B can shuffle A directly to C, and A and C can independently try to path-replace B out of the call. Both events are valid and can co-exist.
QSIG ● QSIG Diversion			QSIG Diversion by rerouting offers advantages over shuffling from a TDM connection: ● After shuffling, the signaling path does not change, and the H.323 trunk resources are not released. ● In scenarios such as call forwarding, if QSIG Diversion by rerouting is successful, it may provide a direct media and signaling connection.
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Table 35: IP hairpinning/shuffling feature interactions ***20 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Russian Intrusion ● Scenario A	Endpoints A and B have a shuffled or hairpinned connection	A public network operator attempts to use Russian intrusion to break into the conversation, and There are no audio channels available on a TN2302AP in either A's or B's network region	The switch sends reorder tone to the public network operator, the same as if A and B had been circuit-switched and a similar attempt failed for lack of switch resources.
Russian Intrusion ● Scenario B	Endpoints A and B have a shuffled or hairpinned connection	A public network operator attempts to use Russian intrusion to break into the conversation, and A and B satisfy the conditions for Russian intrusion, and There is 1 audio channel available for A on a TN2302AP in A's and B's network region	Both A's and B's audio streams are redirected back to a TN2302AP port, and the public network operator connects into the call.
Service Observing ● Scenario A	Endpoints A and B have a shuffled or hairpinned connection	User C attempts to service observe into the conversation, and There are no audio channels available on a TN2302AP in either A's or B's network region	Reorder tone is sent to user C, the same as if A and B had been circuit-switched and a similar attempt failed for lack of switch resources.

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Table 35: IP hairpinning/shuffling feature interactions ***21 of 25***

	Conditions		**Result**
Feature	**If**	**And(s)**	**Then**
Service Observing - Scenario B	Endpoint A and endpoint B have a shuffled audio connection	User C attempts to service observe into the conversation, and There is 1 audio channel available for A on a TN2302AP in A's network region, and there is 1 audio channel available for B on a TN2302AP in B's network region,	The switch redirects the audio stream of both endpoints A and B back to a TN2302AP port, and connects C into the call. The observed parties might hear the 20-300ms break and deduce that they are being observed. We recommend administering service-observed endpoints to prevent shuffling (**IP-IP Direct Audio Connection** field on the **Station** form is n.) For hairpinning, however, the 20-30ms break in conversation should always be short enough to ignore.
Service Observing - Scenario C	Endpoint A and endpoint B have a hairpinned connection	User C attempts to service observe the conversation, and There is 1 audio channel available for A on a TN2302AP in A's network region, and there is 1 audio channel available for B on a TN2302AP in B's network region	The switch redirects the audio stream of both endpoints A and B back to a TN2302AP Prowler port each, and connects C into the call. There is no interaction between Service Observing of a VDN and hairpinning or shuffling (see Call Vectoring).
Terminating Extension Groups			The Terminating Extension Group feature uses simulated bridged appearances to ring multiple endpoints simultaneously. Interactions between shuffling, hairpinning, and bridged appearances are covered in the Bridging section.
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Table 35: IP hairpinning/shuffling feature interactions ***22 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Transfer	Endpoint A and endpoint B have a shuffled or hairpinned connection	User A presses the transfer button,	● If there is 1 audio channel available for A on a TN2302AP in A's network region, the switch redirects the audio stream of endpoint A back to a TN2302AP port. A new call appearance lamp on Set A turns on immediately, and A hears dial tone on the new call appearance. ● If there is 1 audio channel available for B on a TN2302AP in B's network region, and if B is administered for music on hold, the switch redirects the audio stream of endpoint B back to a TN2302AP Prowler port. ● If there are no audio channels available for A on a TN2302AP in A's network region, then a new call appearance on set A flashes. Depending on the type of set that A is using, A might hear a locally-generated tone (such as reorder tone or a "program error" sound). User A can: - Reselect the held call appearance and reconnect to B.
Transfer (Cont.)			- Wait a few seconds and then try another call appearance, should a TN2302AP channel have since become available.
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Table 35: IP hairpinning/shuffling feature interactions ***23 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Transfer - Abort Transfer			If a transfer aborts before completion, endpoints A and B are shuffled or hairpinned back together. Once a transfer completes, the newly-connected parties are hairpinned or shuffled together, if both meet the criteria.
Transfer - Soft Hold			Shuffled or hairpinned connections cannot be set up between two endpoints while either has a TDM party on soft hold awaiting a transfer. However shuffled or hairpinned connection between two endpoints is possible if either endpoint has only IP parties on soft hold awaiting a transfer. For example, IP sets A and B have a shuffled or hairpinned connection. If A presses the transfer button, gets dial tone, and calls set C, then the call from A to C can transition to a shuffled connection. If now A presses the transfer button a second time but there are now no TN2302AP ports available, the transfer attempt fails. There is no lamp to flutter on a conference button, but the line appearance lamp flutters. Endpoints A and C are now talking to each other, and B is still on soft hold.
Transfer - Pull Transfer			If the **Pull Transfer** field on the **System Parameters Features** form is set to y (yes), the called party on a transfer can press the transfer button to complete the transfer. Pull Transfer has the same interactions with hairpinning and shuffling as calling party transfer.
Transfer - Station transfer with callback			A call that is returned to the transferring party by the Station transfer with Callback feature is treated just like an incoming call.

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Table 35: IP hairpinning/shuffling feature interactions ***24 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Transfer ● Transfer upon hangup			Transfer upon hangup has the same interactions with hairpinning and shuffling as normal transfer.
Transfer ● Transfer with misoperation handling			A trunk call that is returned to the transferring party by the misoperation handling feature is treated the same as any other incoming trunk call.
Transfer ● Transfer upon Hold	IP endpoint A has only IP endpoint B on hold	B is not using a TN2302AP port, and User A presses the transfer button	● If there is 1 audio channel available for A on a TN2302AP in A's network region, the switch redirects the audio stream of endpoint A back to a TN2302AP. A new call appearance lamp on set A turns on immediately, and A hears dial tone on the new call appearance. ● If there are no audio channels available for A on a TN2302AP in A's network region, then a new call appearance on set A flashes. Depending on the type of set that A is using, A may hear a locally-generated tone (such as reorder tone or a "program error" sound). At this point user A may choose to - Reselect the held call appearance and reconnected to B - Wait a few seconds and then try another call appearance in the hope that a TN2302AP audio channel has since become available.
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Table 35: IP hairpinning/shuffling feature interactions ***25 of 25***

	Conditions		Result
Feature	**If**	**And(s)**	**Then**
Whisper Page ● Scenario A	Endpoint A and endpoint B have a shuffled or hairpinned connection	A third party, C, attempts to use whisper page to talk to user A, and There are no audio channels available on a TN2302AP in A's network region, or there are no audio channels available on a TN2302AP in B's network region,	The switch sends reorder tone and lamp flashes to user C, the same as if A and B had been circuit-switched and a similar attempt failed for lack of switch resources.
Whisper Page ● Scenario B	Endpoint A and endpoint B have a shuffled or hairpinned connection	A third party, C, attempts to use whisper page to talk to user A, and There is 1 audio channel available for A on a TN2302AP in A's network region, and there is 1 audio channel available for B on a TN2302AP in B's network region	The switch redirects the audio stream of both A and B back to a TN2302AP port and connects C to A. User B may deduce from the 20-300ms break that user A just received a whisper page. To avoid this possibility, administer endpoints that participate in whisper paging to prevent shuffling (**IP-IP Direct Audio Connection** field on the **Station** form is n.) For hairpinning, however, the 20-30ms break in conversation should always be short enough to ignore.

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Fax, TTY, and modem over IP feature interactions

- Call Admission Control

  Call Admission Control (CAC) uses the administered values when considering bandwidth availability for a specific call and allows or denies the call accordingly. However, when a call originates as an audio call and then changes to a digitized FAX or modem call, CAC may allow the call due to resource availability, but then deny the call because of the additional bandwidth necessary.

- Modular Messaging

  Modular Messaging supports fax and TTY transport over IP networks

- IP Office

  IP Office supports fax and TTY transport over IP networks

- Multi-Tech MultiVOIP

  The T.38 FAX capability works with Multi-Tech MultiVOIP.

- Call Detail Recording (CDR)

  You can send call records for FAX, modem, or TTY calls over an IP network to a CDR device. Call records are recorded in the same way as any other calls over an IP network, depending on trunk group administration.

- Conference/Transfer/Hold

  For fax and modem calls over the IP network, failures will occur (that is, data will be lost) when a call is placed on hold (for the Hold, Conference, or Transfer features). In addition, failures will occur (that is, data will be lost) as a result of a conference that causes the master gateway of a call to change in the middle of a call.

- Call Management System (CMS)

  Originating or terminating endpoints or trunk groups that are involved in a FAX, modem or TTY call can be measured by CMS.

- Interactive Response

  Interactive Response does not support FAX and modem calls received over an IP network. Interactive Response treats TTY calls sent over an IP network in the same way it treats TTY calls arriving over any other trunk facility.

- Shuffling / Direct IP Connections

  Audio Shuffling and Direct IP Connections are supported on fax, modem, and TTY calls.

Appendix A: Using IP Routes

Using IP routes

On LANs that connect to other networks or subnetworks, Avaya recommends that you define a default gateway. See Defining a LAN default gateway on page 50 for more information. Only in rare cases should you add other routes to define specific network paths through other gateways.

Note:

> Avaya recommends that routing is defined on your data network, rather than through Communication Manager. This section should only be used under exceptional circumstances.

This table describes the network configurations that require explicit IP routes:

Connection Type	Use IP routes when:
Ethernet	● You want the local node to communicate to a remote subnet without routing through the default gateway. ● You want the local node to communicate with any node in a remote network but not with nodes on other networks (this is a network route type).
ppp	● There are one or more intermediate nodes between endpoints.

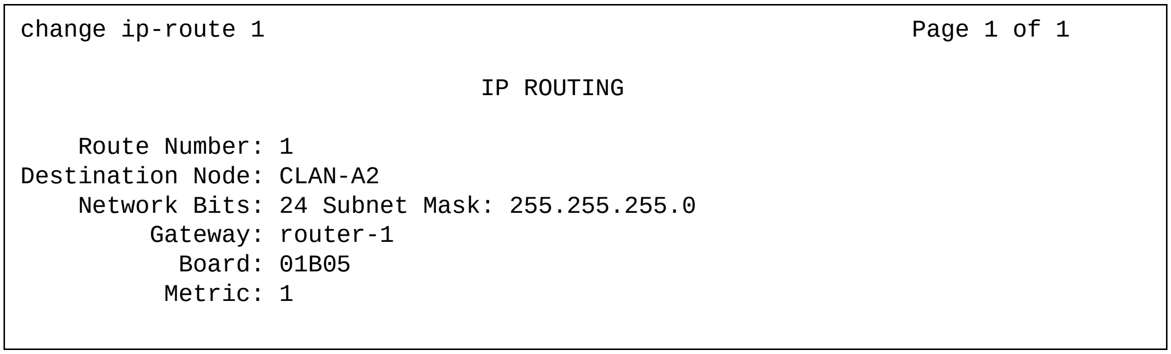
Setting up IP routing

To set up an IP route:

1. Type **`change ip-route`** ***`number`*** and press **Enter**

   where ***`number`*** is the number of the next available IP route. The **IP Routing** screen displays.

IP Routing screen



```
change ip-route 1                                              Page 1 of 1

                                IP ROUTING

    Route Number: 1
Destination Node: CLAN-A2
    Network Bits: 24 Subnet Mask: 255.255.255.0
         Gateway: router-1
           Board: 01B05
          Metric: 1
```

2. Enter the node names for the **Destination** and the **Gateway**, and enter the slot location of the C-LAN (**Board**) on the local switch.

   The destination and gateway node names and their associated IP addresses must be specified on the **Node Names** screen.

The **Route Type** is a display-only field that appears on the screen for the **`display, list,`** and **`change ip-route`** commands. This field indicates whether the route is a *`host`* or *`network`* route. It is a host route if the destination address (associated with the **Destination Node** on the **Node Names** screen) is the address of a single host, or node. It is a network route if the destination address is the address of a network, not a single node.

Advanced IP routing

If you wanted the local C-LAN node to be able to communicate, for example, with the nodes on the 192.168.1.64 subnetwork and not with others, you could do the following:

To set up subnetwork IP routing:P

1. Leave blank the **Gateway Address** field on the **IP Interfaces** screen.
2. Enter a node name — for example, **subnet-1** — and the IP address, **192.168.1.64**, on the **Node Names** screen.
3. Set up an IP route with **subnet-1** in the **Destination Node** field.

IP route example: PPP connections

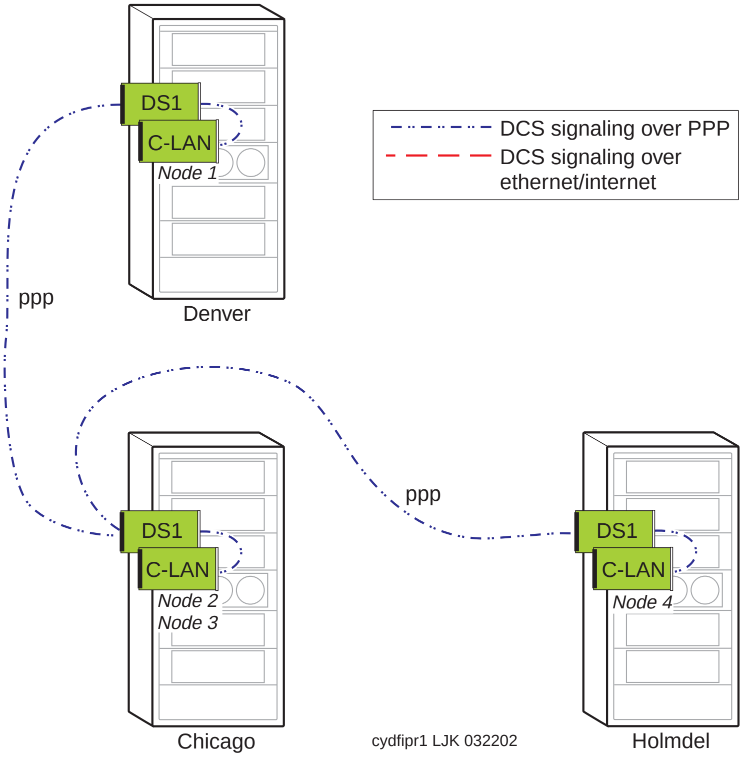
Figure 15 shows a DCS network with PPP signaling connections between systems in Chicago and Denver, and between systems in Chicago and Holmdel. PPP data modules are administered between IP node 1 and IP node 2 on Chicago and Denver, and between IP nodes 3 and 4 on Chicago and Holmdel.

Note:

All nodes in this description, in the diagram, and in the following tables are IP Nodes, and are not DCS switch nodes.

With these connections, Chicago can communicate with Denver and Holmdel without using the IP Routing screen to administer explicit host IP routes. However, Denver and Holmdel need host IP routes to communicate with each other because they are not directly connected.

Figure 15: DCS network with PPP signaling



Note:

The IP routes between nodes for this example are listed in Table 36. The **Destination Node** and **Gateway Node** columns in the table show the nodes to enter on the **IP Routing** screen to administer a host IP route. On the **IP Routing** screen, enter the node names assigned on the **Node Names** screen for these nodes.

Table 36: IP routes between nodes for PPP example

System location	IP Node connections	Destination node	Gateway node	Route type	Comments
Denver	1 —> 4	4	2	host	IP route needed because there is an intermediate node between nodes 1 & 4.
Holmdel	4 —> 1	1	3	host	IP route needed because there is an intermediate node between nodes 4 & 1.

Note:

The PPP data modules on systems Denver and Holmdel for the connections to Chicago must be enabled before the IP routes can be administered.

Note:

Nodes 2 and 3 in this example are two ports on the same C-LAN board. Messages from node 1 destined for node 4 arrive at node 2; the C-LAN ARP software routes the messages to node 4 through node 3.

IP route example: PPP with Ethernet Connections

Figure 16 shows two interconnected (sub)networks. There are three systems in a DCS network with a PPP signaling connection between systems in Chicago and Denver and an Ethernet signaling connection between the system in Chicago and the adjunct. Chicago and Denver and the adjunct are on one (sub)network and Holmdel is on another (sub)network.

Note:

All nodes in this description, in the diagram, and in the following tables are IP Nodes, and are not DCS switch nodes.

Chicago acts as a gateway to convert between the two signaling protocols. PPP data modules are administered between nodes 1 and 3 on Chicago and Denver, and Ethernet data modules are administered on Chicago and Holmdel for the C-LAN Ethernet port interfaces to their LANs. With these connections, Chicago can communicate with Denver and with the adjunct without using the IP Routing screen to administer explicit IP routes.

Normally, node 5 is defined as the default gateway for node 2 on the **IP Interfaces** screen, which enables Chicago to communicate with Holmdel without an explicit IP route defined. However, if node 5 is not assigned as the default gateway for node 2, Chicago needs an IP route to communicate with Holmdel because these systems are on different (sub)networks. Node 6 is normally defined as the default gateway for node 7. If it is not, Holmdel needs an IP route to communicate with Chicago.

Also, Denver needs an IP route to communicate with Holmdel, because Denver is connected to Chicago using PPP and there are intermediate nodes between Denver & Holmdel.

Figure 16: Network with PPP and Ethernet connections

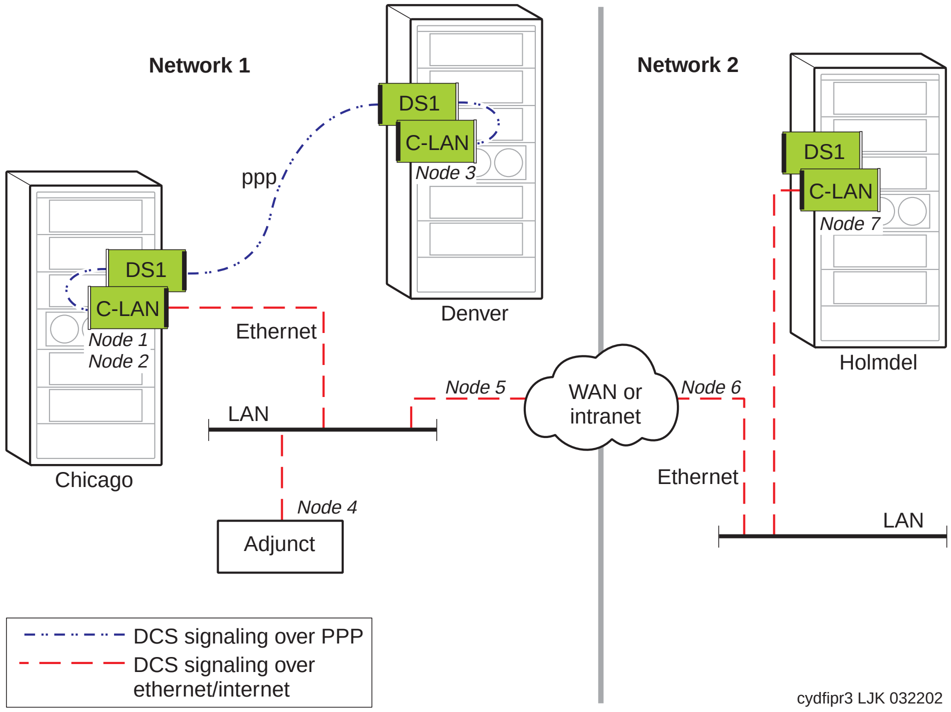


Table 37 shows the IP routes needed if nodes 5 and 6 are not defined as default gateways for nodes 2 and 7.

Table 37: IP route examples: PPP-ethernet example ***1 of 2***

System location	IP Node connections	IP Route Destination node	IP Route Gateway node	Comments
Chicago	2 —> 7	7	5	IP route needed because nodes 2 & 7 are on different subnets and the **Gateway Address** field for the node-2 C-LAN is blank on the **IP Interfaces** screen.
Denver	3 —> 4	4	1	IP route needed because 3 is connected to 1 using PPP and there are intermediate nodes between 3 & 4. The data module for the PPP connection between nodes 3 and 1 must be enabled before administering this route.
	3 —> 7	7	1	IP route needed to because 3 is connected to 1 using PPP and there are intermediate nodes between 3 & 7. The data module for the PPP connection between nodes 3 and 1 must be enabled before administering this route.
				1 of 2

Table 37: IP route examples: PPP-ethernet example *2 of 2*

System location	IP Node connections	IP Route Destination node	IP Route Gateway node	Comments
Holmdel	7 —> 4	4	6	IP route needed because nodes 4 & 7 are on different subnets and the **Gateway Address** field for the node-7 C-LAN is blank on the **IP Interfaces** screen.
	7 —> 2	2	6	IP route needed because nodes 2 & 7 are on different subnets and the **Gateway Address** field for the node-7 C-LAN is blank on the **IP Interfaces** screen.
	7 —> 3	3	2	IP route needed because nodes 3 & 7 are on different subnets. This route depends on route 7—>2. Note: this route is not be needed if node 6 is administered for proxy ARP to act as a proxy agent for node 3.

2 of 2

IP route example: Ethernet-only connections

Figure 17 shows three interconnected (sub)networks. There are three systems in a DCS network with Ethernet signaling connections between them. The systems in Chicago and Denver and the adjunct are on one (sub)network and Holmdel is on another (sub)network. Nodes 1, 2, and 6 are C-LAN ports. Node 3 is the adjunct interface port to the LAN. Nodes 4, 5, and 7 are interfaces to the WAN/Internet cloud and have IP addresses that are on different (sub)networks. An Ethernet data module and IP Interface is administered for the C-LAN Ethernet port on each system.

Note:

All nodes in this description, in the diagram, and in the following tables are IP Nodes, and are not DCS switch nodes.

Chicago and Denver can communicate with each other and with the adjunct without using the **IP Routing** screen to explicitly administer host IP routes. Normally, node 4 is defined as the Gateway Address for node 1 on the **IP Interfaces** screen, which enables Chicago to communicate with Holmdel without an explicit host IP route defined. However, if node 4 is not assigned as the **Gateway Address** for node 1, Chicago needs an IP route to communicate with Holmdel because these systems are on different (sub)networks. Similarly, node 5 is normally defined as the default gateway for node 6. If it is not, Holmdel needs an IP route to communicate with Chicago.

In this configuration, network IP routes can be used alone, or in combination with host IP routes, to tailor access among nodes. For example, if you want node 1 to be able to communicate with any node on (sub)networks 2 and 3, define node 4 as the **Gateway Address** for node 1. Then you do not need any IP routes defined for node 1. If you want node 1 to be able to communicate with all nodes on (sub)network 3 but none on (sub)network 2, define a network IP route to (sub)network 3 (and *not* assign node 4 as the Gateway Address for node 1). Then node 1 can communicate with any node on (sub)network 3 without defining host IP routes to them.

Figure 17: Network with ethernet-only connections

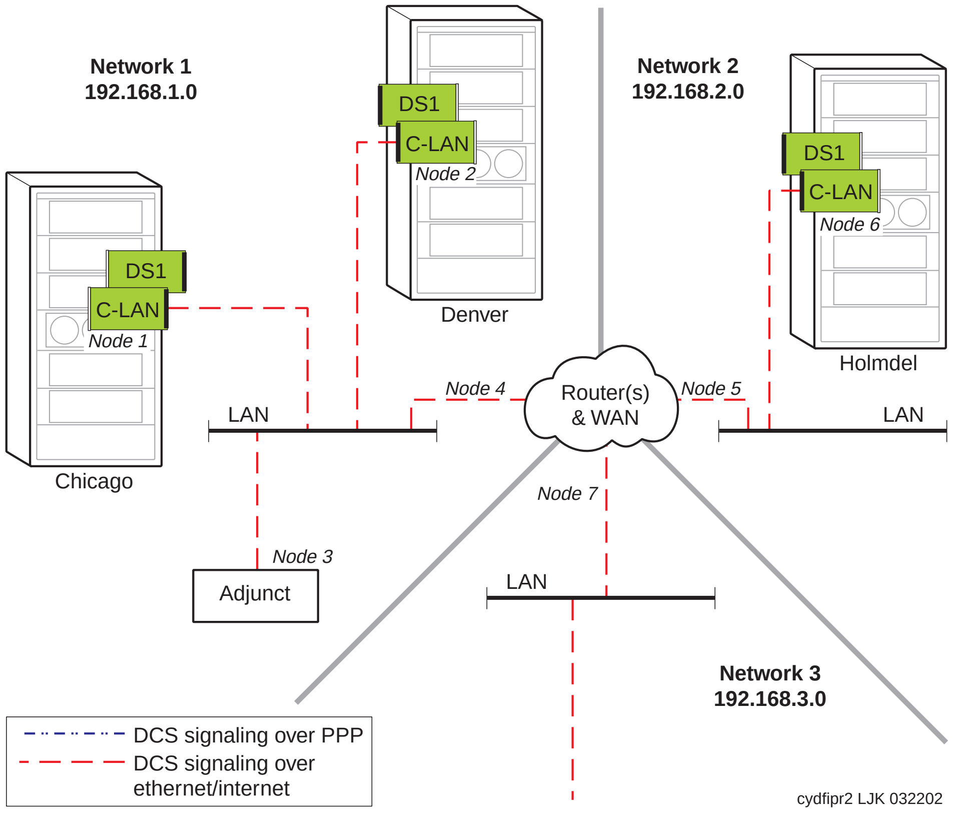


Table 38 shows the IP routes if node 4 is not defined as the **Gateway Address** (on the **IP Interfaces** screen) for nodes 1, 2, and 3, but node 5 is defined as the **Gateway Address** for node 6.

Table 38: Ethernet-only IP route examples (if node 4 is not defined)

System location	IP Node connections	IP Route Destination node	IP Route Gateway node	Route type	Comments
Chicago	1 —> 6	6	4	host	IP route needed because nodes 1 & 6 are on different subnets and no **Gateway Address** is specified for the node-1 C-LAN on the **IP Interfaces** screen.
	1—> network 3	network-3	4	network	This route enables node 1 to communicate with any node on Network 3. Associate the node name **network-3** with the IP address **192.168.3.0** on the **Node Names** screen.
Denver	2 —> 6	6	4	host	IP route needed because nodes 2 & 6 are on different subnets and no **Gateway Address** is specified for the node-1 C-LAN on the **IP Interfaces** screen.
Holmdel					No IP routes are needed on system Holmdel because node 5 is defined as the **Gateway Address** for node 6.

Appendix B: Internet Control Message Protocol (ICMP) ECHO messages

Servers running Communication Manager, Avaya gateways, and Avaya IP telephones use Internet Control Message Protocol (ICMP) ECHO messages (also called pings) continuously to assess the availability of the IP network. Table 39: Ping usage and consequences of ping failure is a current listing of when and why the pings are used, and the consequences of ping failures caused by real network outages or ICMP message filtering or suppression.

Table 39: Ping usage and consequences of ping failure ***1 of 3***

Function	Ping Originator	Ping Destination	Purpose	Consequence of Ping Failure
IP Trunk Bypass	MEDPRO/ TN2302	Far-end signaling endpoint (for example, a CLAN)	Periodic pings are sent to measure performance characteristics of connectivity to the far end. Rate is administrable on the ***change system-parameters ip-options*** form.	If the ping results are not satisfactory, the IP trunk is put into by-pass mode, which means that it will not be used for new calls. Satisfactory is defined on the ***change system-parameters ip-options*** form.
IP Trunk Signaling Link Connectivity	CLAN/ TN799 or S8300 (Directly from server)	Far-end signaling endpoint (CLAN, S8300, or other gateway, gatekeeper, or SIP proxy	Periodic pings are sent to determine if the far end is reachable. The period depends on how busy the system is, but is no more frequent than once every 15 minutes.	If the ping fails, the IP trunk is taken out of service and calls on it are dropped.
Inter-Network Region Connectivity	G700/G350	G700/G350 (in another network region)	Pings are periodically sent to test connectivity of inter-connected network regions. Gateways and telephones are administratively assigned to network regions.	A Warning Alarm is generated. Warnings do not usually result in a call to INADS.

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Table 39: Ping usage and consequences of ping failure ***2 of 3***

Function	Ping Originator	Ping Destination	Purpose	Consequence of Ping Failure
IP Telephone Reachability	CLAN/ TN799 or S8300 (Directly from server)	IP station	Pings are periodically sent to determine if the IP station is reachable.	A Warning Alarm is generated. Warnings do not usually result in a call to INADS.
Servers and Gateways Connectivity	CLAN/ TN799 VAL/ TN2501	The default gateway and up to two other IP addresses chosen from the list of known addresses on the same subnet (specified in the ***change node-names ip*** form)	A periodic test to determine if the server or gateway has IP connectivity to the outside world.	If all pings fail two consecutive times, a Warning Alarm is generated. Warnings do not usually result in a call to INADS.
	TN2302	The default gateway for this circuit pack	A periodic test to determine if the circuit pack has IP connectivity to the outside world	The test fails when run manually. There is no alarm generated.
	Other servers, including S8700, S8500, S8300 IPSI/ TN2312	Most circuit packs and servers have manual ping capabilities, which are not required or used during normal operations. The ping is used during manual network troubleshooting.		None

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Table 39: Ping usage and consequences of ping failure ***3 of 3***

Function	Ping Originator	Ping Destination	Purpose	Consequence of Ping Failure
IP Telephones	IP Softphone	Far end of voice conversation (either an IP station or a MEDPRO resource on a gateway)	Used to determine the round-trip time when RTCP is disabled (for display purposes only)	None
	IP Station (not Softphone)	The IP station's default gateway	If the station's signaling link has been down for one hour (no response to Gatekeeper Requests (GRQs)) and this test fails, the IP station assumes insanity in its network stack/interface.	The IP station reboots in order to resolve the assumed local problem.

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