Administration for Network Connectivity for Avaya MultiVantage™ Software

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Notice

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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.

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

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

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

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

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The final responsibility for securing both this system and its networked equipment rests with you - Avaya's customer system administrator, your telecommunications peers, and your managers. Base the fulfillment of your responsibility on acquired knowledge and resources from a variety of sources including but not limited to:
- Installation documents
- System administration documents
- Security documents
- Hardware-/software-based security tools
- Shared information between you and your peers
- Telecommunications security experts

To prevent intrusions to your telecommunications equipment, you and your peers should carefully program and configure:
- Your Avaya-provided telecommunications systems and their interfaces
- Your Avaya-provided software applications, as well as their underlying hardware/software platforms and interfaces
- Any other equipment networked to your Avaya products.

Voice Over Internet Protocol (VoIP)

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

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Avaya Inc. is not responsible for any radio or television interference caused by unauthorized modifications of this equipment or the substitution or attachment of connecting cables and equipment other than those specified by Avaya Inc. The correction of interference caused by such unauthorized modifications, substitution or attachment will be the responsibility of the user. Pursuant to Part 15 of the Federal Communications Commission (FCC) Rules, the user is cautioned that changes or modifications not expressly approved by Avaya Inc. could void the user's authority to operate this equipment.
The equipment described in this manual complies with standards of the following organizations and laws, as applicable:
- Australian Communications Agency (ACA)
- American National Standards Institute (ANSI)
- Canadian Standards Association (CSA)
- Committee for European Electrotechnical Standardization (CENELEC) – European Norms (EN's)
- Digital Private Network Signaling System (DPNSS)
- European Computer Manufacturers Association (ECMA)
- European Telecommunications Standards Institute (ETSI)
- FCC Rules Parts 15 and 68
- International Electrotechnical Commission (IEC)
- International Special Committee on Radio Interference (CISPR)
- International Telecommunications Union - Telephony (ITU-T)
- ISDN PBX Network Specification (IPNS)
- National ISDN-1
- National ISDN-2
- Underwriters Laboratories (UL)

Product Safety Standards

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

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

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

Electromagnetic Compatibility (EMC) Standards

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

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

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

Federal Communications Commission Statement

Part 15:

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

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

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

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

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

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

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

Means of Connection

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

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

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

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

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

It is recommended that repairs be performed by Avaya certified technicians.
The equipment cannot be used on public coin phone service provided by the telephone company. Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.
This equipment, if it uses a telephone receiver, is hearing aid compatible.

Canadian Department of Communications (DOC) Interference Information

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

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

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

DECLARATIONS OF CONFORMITY

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

Avaya Inc. in the United States of America hereby certifies that the equipment described in this document and bearing a TIA TSB-168 label identification number complies with the FCC's Rules and Regulations 47 CFR Part 68, and the Administrative Council on Terminal Attachments (ACTA) adopted technical criteria.
Avaya further asserts that Avaya handset-equipped terminal equipment described in this document complies with Paragraph 68.316 of the FCC Rules and Regulations defining Hearing Aid Compatibility and is deemed compatible with hearing aids.
Copies of SDoCs signed by the Responsible Party in the U. S. can be obtained by contacting your local sales representative and are available on the following Web site:

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

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

http://www.part68.org/

by conducting a search using "Avaya" as manufacturer.

European Union Declarations of Conformity

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). This equipment has been certified to meet CTR3 Basic Rate Interface (BRI) and CTR4 Primary Rate Interface (PRI) and subsets thereof in CTR12 and CTR13, as applicable.

Copies of these Declarations of Conformity (DoCs) signed by the Vice President of MultiVantage™ Solutions research and development, Avaya Inc., can be obtained by contacting your local sales representative and are available on the following Web site:
http://support.avaya.com/elmodocs2/DoC/IDoC/index.jhtml/

Japan

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

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Network Connections

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

Belgium	Germany	Luxembourg
Netherlands	Spain	United Kingdom

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

- Maximum power output –5 dBm to -8 dBm
- Center Wavelength 1310 nm to 1360 nm
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Contents

Contents

Contents

Contents

About this Document

This book describes how to administer connections among Avaya MultiVantage solutions. The main focus is on TCP/IP for DCS signaling and H.323 trunks.

Purpose

This document provides information about connections between Avaya MultiVantage solutions in a network using IP connections.

For installation or upgrade procedures for establishing physical connectivity between switches, or for connecting the CMS and Intuity AUDIX adjuncts to a switch. Refer to the upgrades and installation documents listed in the References section.

Audience

This document is intended for people involved in planning, designing, or administering networks using IP connectivity.

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, "Circuit Packs" provides an overview of circuit packs used with IP connectivity.
- Chapter 2, "Circuit Packs" provides procedures for initial administration of IP trunks using H.323 IP connections.
- Chapter 4, "Quality of Service and Voice Quality Administration" provides instructions for administering Quality of Service on an Avaya MultiVantage solution.
- Chapter 5, "QSIG" describes supported QSIG features and discusses compatibility issues for setup.
- Chapter 6, "A Complex Networking Example" shows administration screens used for setting up an example of a complex network.
- Appendix A, "Private Networking" describes DCS features and QSIG.
- Appendix B, "IP Routes" explains how to add IP routes when you want to define specific network paths through gateways other than the default gateway.
- Appendix C, "IP Addressing" describes IP addressing, subnetting, and routing.
- Index

NOTE:
The Networking Examples chapter has been moved to a separate document that can be found on the Avaya MultiVantage documentation CD.

Conventions used

This section explains how certain terms are used in this book.

- Screen

  A screen is the set of switch-administration interface pictures that contain the fields that hold the switch-translations values. In some parts of this book, the terms “screen” and “form” are used interchangeably.

- Node

  In this book, node is used to refer to a network interface, as with TCP/IP connectivity. For example, each of the 17 ports on the C-LAN board is a node. This is also the common usage in a data networking environment. With these definitions, a “DCS node” (a switch) can have many “IP nodes,” (network interfaces). This is how the term is used on the Node Names, Data Module, Processor Channel, and IP Routing screens.

  There is another meaning for the term “node” for Avaya MultiVantage solutions connected in a network. In a DCS network, node means a switch or adjunct.

- IP Media Processor

  IP Media Processor refers to the TN2302 circuit pack.

- IP Interface and MedPro

  In this book, IP Interface assembly, MedPro board, and the TN802B circuit pack are used interchangeably.

  The official name for the TN802B circuit pack is the IP Interface assembly. It is a media processing circuit pack in a 3-slot wide assembly. It can be administered to operate in either IP Trunk mode or MedPro mode.

How to access this book from the web

If you have internet access, you can view and download the latest version of *Administration for Network Connectivity*. To view the book, you must have a copy of Adobe Acrobat Reader (www.adobe.com).

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Networking Overview

1

This chapter provides background information to help you understand and use this book. It describes how Avaya MultiVantage solutions can be connected, with a focus on IP connectivity.

Avaya MultiVantage solutions Connectivity

This section describes the basic components of a network of switches, and how voice and signaling data are transmitted between switches for different types of switch connections. It also provides a summary of the administration procedures for connecting switches via an IP network using the C-LAN and media processor circuit packs.

Connectivity Overview

Avaya MultiVantage solutions can be connected 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.

Types of connections

Trunks

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, which provide connections to the public telephone network through a central office, and tie trunks, which provide connections between switches in a private network.

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

The H.323 trunk allows voice and fax data to be transmitted over the Internet to another system with H.323 trunk capability. The H.323 trunk supports Q.931 services such as DCS+ and QSIG.

Networks

When two or more switches are connected via tie trunks, they form a private network. There are two basic types of networks for Avaya switches:

- Main-satellite/tributary (MS/T) — A network of switches in which a main switch is fully functional and provides attendants and CO trunks for connected satellite switches. Tributary switches are connected to the main and may have their own attendant and CO trunks. The main switch may be connected to one or more Electronic tandem networks (ETNs).
- Electronic tandem network (ETN) — A wide-area network of switches in which a call can tandem through one or more switches on its way from the originating switch to the destination switch. ETNs have a uniform dial plan (UDP), automatic alternate routing (AAR), and automatic route selection (ARS).

AT&T provides a service called software-defined network (SDN) that allows you to build a private network through the AT&T public network facilities. An ETN can be combined with an SDN to form a hybrid (ETN/SDN) network.

MCI provides a service called N-Quest that allows you to build a private network through the AT&T public network facilities.

The switches in MS/T or ETN networks need to be provisioned with special networking software packages.

DCS

Distributed Communications System (DCS) is a messaging overlay for ETN or MS/T networks. The overlay provides signaling connections between network nodes that enable certain call features to operate transparently across the DCS network. The transparent features operate as if the switches in the DCS network were a single switch. For example, the DCS Call Coverage feature enables calls to an extension on one switch to be covered by extensions on a remote switch in the network.

DCS consists of two components, routing and message signaling. Routing the message requires one of several networking software packages. Uniform Dial Plan is required and is included with DCS at no additional charge.

Although DCS is actually a messaging overlay for an existing network, it is commonly thought of as a type of network itself. In this document, DCS network refers to a cluster of switches that are part of an existing ETN or MS/T network and are administered for DCS.

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 three ways:

- over a processor interface (PI) channel (on the si model) or a packet gateway (PGATE) channel (r model) using the X.25 protocol
- over an ISDN-PRI D-channel (all models)
- over a TCP/IP (either PPP or 10/100Base-T Ethernet) connection (all models)

X.25 connections are not supported in newer systems. Instead, Avaya MultiVantage provides TCP/IP connections or ISDN-PRI for DCS signaling. However, existing systems with X.25 and/or ISDN-PRI DCS signaling can be upgraded to the latest version and keep those signaling links, or a new system can be added to an existing DCS network. Connections to the CMS Call Center and Intuity AUDIX adjuncts can use either X.25 or 10/100Base-T DCS signaling.

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 and the gateway enables 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.

What is transmitted between connected switches?

A telephone call consists of voice (bearer) data and call-signaling data. If the call is over a DCS network, DCS signaling data is also required. The DCS signaling data is sent over a separate path from the voice and call-signaling data.

Call-signaling data

The call-signaling data includes messages that set up the call connection, maintain the connection during the call, and remove the connection when the call is finished.

DCS-signaling data

The DCS-signaling data is separate from the call-signaling data. The type of connection determines the signaling protocol for the DCS-signaling data transmission.

Transmission between switches

The following table gives a summary of the different types of call connections and how the voice and signaling data are transmitted between switches.

Table 1. Call connections and data transmissions

Connection Type	Tie Trunk		LAN or WAN	
	Voice & Call-Signaling	DCS Signaling	Voice	Call & DCS Signaling
DCS+ over ISDN	T1/E1 facilities, ISDN-PRI B-Channel	TSCs on the ISDN-PRI D-Channel		TSCs on the ISDN-PRI D-Channel
QSIG over ISDN	T1/E1 facilities, ISDN-PRI B-Channel	CISCs on the ISDN-PRI D-Channel		CISCs on the ISDN-PRI D-Channel
X.25	T1/E1 facilities using ISDN-PRI or DS1 B-Channel OR Analog trunk	Packet PVC		
C-LAN PPP	T1/E1 facilities using ISDN-PRI or DS1 B-Channel OR Analog trunk	Packet PVC		
C-LAN Ethernet	T1/E1 facilities using ISDN-PRI or DS1 B-Channel OR Analog trunk			TCP Packets on processor channels (DCS signaling only)
IP Interface			RTP Packet IP media processor	TCP Packets over signaling groups (C-LAN)

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

For C-LAN Ethernet connections, the signaling and voice data are sent together over tie-trunk facilities as TDM-multiplexed frames. The DCS signaling data is sent as TCP datagrams over an IP network through the C-LAN.

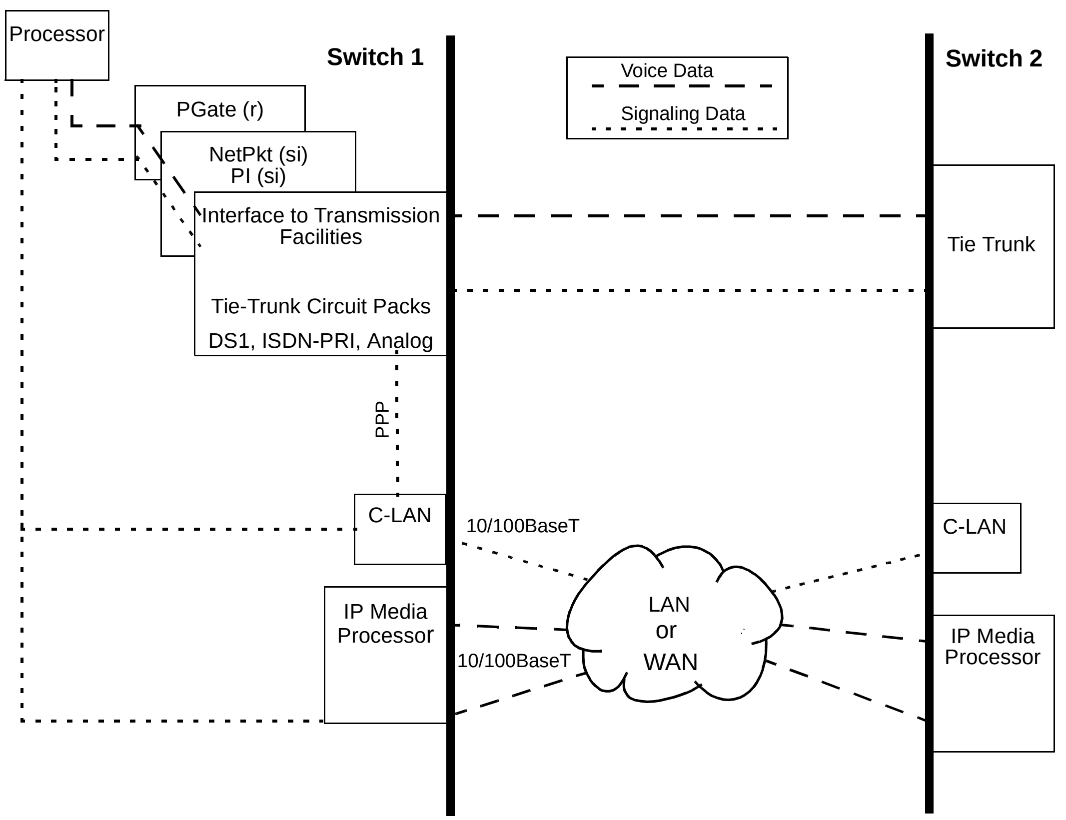


Figure 1. Components of Switch Connectivity

Types of circuit packs

The following section describes the function of each circuit pack shown in the Components of Switch Connectivity figure.

Processor

The processor board is the main control element in handling the call. This is the UN331 or UN332 for the r model, the TN2404 for the si model, and the TN2402 for the csi model.

PGATE (r only)

On the r model, the PGATE board (TN577) connects the processor to the packet bus and terminates X.25 signaling.

NetPkt (si only)

The TN2401 Net/Pkt interface circuit pack provides the network control interface (NETCON), the packet interface (PACCON), and, if BX.25 connectivity is not required, the processor interface (PI). The TN2401 provides eight asynchronous data channels. The TN2401 does not include modems. The TN2401 is required for the si model to save translations to the 5-volt ATA flash memory card.

C-LAN

The C-LAN circuit pack (TN799) enables signaling data to be transmitted via the TCP/IP protocols across a LAN or WAN. Signaling types include call setup and teardown, registration of IP softphones, TSCs, QSIG and DCS signaling over IP trunks.

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 — either via an Ethernet LAN or a point to point protocol (PPP) connection — depending on how the data link is administered. If the link is administered for an Ethernet connection, the signaling data is sent out on a 10/100Base-T network, which is connected directly to the C-LAN Ethernet port. If the link is administered for a PPP connection, C-LAN inserts the signaling data on the TDM bus for subsequent inclusion (via the switching fabric) in the same DS1 bit stream as the voice transmissions.

The C-LAN board can be inserted in any available port slot.

IP Media Processor

The TN2302 IP Media Processor reduces per-port costs and improves quality through its dynamic jitter buffers. It performs echo cancellation, silence suppression, DTMF detection, and conferencing.

The TN2302 IP Media Processor can work in the same server with the TN802 IP Trunking and the TN802B IP Interface Assembly. The software chooses media processing resources for an IP endpoint from the TN2302 over the TN802B, when both type of media processing are available on the system.

The TN2302 enables the transmission of voice and fax data (non-DCS signaling) over IP connections. The IP Media Processor is suitable for H.323 multimedia applications in Avaya MultiVantage solutions and other H. 323 V2 compliant endpoints.

IP-Interface

Like the TN2302 IP Media Processor, 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 enables support of applications that comply with the H.323-v2 protocols. It also operates in the IP Trunk mode supporting R7 IP trunks that emulate DS1 connections, however this method of connection is obsolete.

IP Server Interface

The TN2312 IP Server Interface circuit pack (IPSI) provides an interface between the S8700 Media Server and its port networks. The IPSI connects to the S8700 Media Server via Ethernet. One IPSI can support up to 5 port networks.

Connection types

This section gives an overview of the types of connections that can be set up with Avaya MultiVantage solutions, and capacities for some connectivity parameters.

Types of connections

The following table lists the types of connections possible with each model and adjunct.

Table 2. Types of connections

Server Type	Connection Type	Endpoint or Service
R9csi and later	Ethernet	DCS, CMS, Intuity AUDIX, IP Telephone, IP Softphone, Avaya R300
	Synchronous PPP	DCS
	ISDN-PRI	DCS+, QSIG
	H.323 Trunk	DCS+, QSIG

Continued on next page

Table 2. Types of connections *(Continued)*

R9si and later	Ethernet	DCS, CMS, Intuity AUDIX, IP Telephone, IP Softphone, Avaya R300
	Synchronous PPP	DCS
	ISDN-PRI	DCS+, QSIG
	BX.25	DCS, CMS, Intuity AUDIX, DEFINITY AUDIX
	H.323 Trunk	DCS+, QSIG
R9r and later	Ethernet	DCS, CMS, Intuity AUDIX, IP Telephone, IP Softphone, Avaya R300
	Synchronous PPP	DCS
	ISDN-PRI	DCS+, QSIG
	BX.25	DCS, CMS, Intuity AUDIX, DEFINITY AUDIX
	H.323 Trunk	DCS+, QSIG

If an R9 or later server is connected to two endpoints by different connection types, it acts as a gateway (protocol converter) between the endpoints.

Avaya R300 Remote Office Communicator

The Avaya R300 Remote Office Communicator provides an effective way to maintain remote DCP and analog phones and local trunks from an Avaya MultiVantage solution. In addition to voice telephony features, the Avaya R300 provides the remote site with data integration capability and provides a conversion of voice and data applications in the same product.

This remote application provides full Avaya MultiVantage functionality and features to the remote site either through a WAN or LAN using IP protocol.

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 architecture of an Avaya MultiVantage solution prevents access to the customer's LAN as depicted in Figure 2.

Figure 2 shows a high level view of the system architecture of a switch with a C-LAN port board. This architectural view shows that logins via the INADS line terminate in software; software communicates with firmware over an internal bus via a limited message set. There are two main reasons why a user can not access a customer's LAN via the INADS line:

- A user logging into software can not get direct access to the C-LAN Firmware. The user can only enter SAT commands that request C-LAN information or to configure C-LAN (IP addresses, clock).
- The C-LAN application TFTP is currently disabled and can not be enabled by Avaya MultiVantage software. TELNET only interconnects C-LAN Ethernet clients to the system management application on the switch. FTP exists only as a server and is used only for firmware downloads, and it cannot connect to the client network.

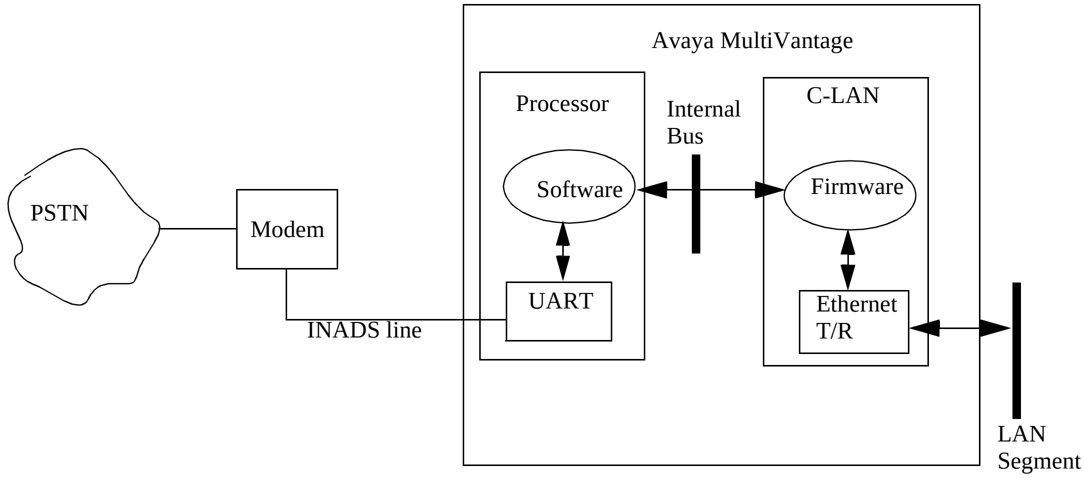


Figure 2. High Level System Architecture

Circuit Packs

2

This chapter contains a description of the TN799 C-LAN, the TN2302 IP Media Processor, the TN2312 IP Server Interface and the UDS1 tie trunk circuit packs. Also included are installation and administration for these circuit packs.

TN799 C-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 TN799 C-LAN circuit pack.

The C-LAN handles the data-link signaling information in one of two configurations; either Ethernet or point-to-point (PPP).

- In the Ethernet configuration, the C-LAN passes the signaling information over a separate TCP/IP network, usually via 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 as described below.
- 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

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

Beginning with the TN799DP, the CLAN supports both Classless Inter-domain Routing and Variable-Length Subnet Masks. These addressing techniques provide greater flexibility in addressing and routing than classful addressing alone.

Installing the TN799 C-LAN

Before you install the C-LAN circuit pack, be sure you understand the requirements of your LAN. Refer to http://support.avaya.com and search for the white paper *Avaya IP Voice Quality Network Requirements*.

The following steps describe installation for the TN799 C-LAN.

Insert C-LAN Circuit Packs

TCP/IP connections (Ethernet or PPP) require a TN799 C-LAN circuit pack, unless your system has embedded Ethernet capabilities. Complete the following steps to install these circuit packs.

1. Determine the carrier/slot assignments of the circuit packs to be added. The C-LAN circuit pack can go into any port slot.
2. Insert the circuit packs into the slots specified in step 1. You do not need to power down the cabinet to install a C-LAN circuit pack.

Administer the C-LAN Bus Bridge for csi models

For the csi model only, complete the following steps to administer the Bus Bridge for the new C-LAN circuit pack. Maintenance parameters can only be changed by an Avaya representative using the craft or higher login.

NOTE:
If there are 2 C-LAN circuit packs installed in this csi switch, administer the Bus Bridge for *only one* of them.

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.

Test the Packet Bus and C-LAN Circuit Pack

To test the packet bus and the TN799 C-LAN circuit pack, the cabinet needs a TN771D Maintenance/Test 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 the maintenance book.
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 TN799 circuit pack, remove it from the cabinet.

Install C-LAN Cables

Install a cable from the backplane peripheral connector to a hub or Ethernet switch as required. See the figure below.

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.

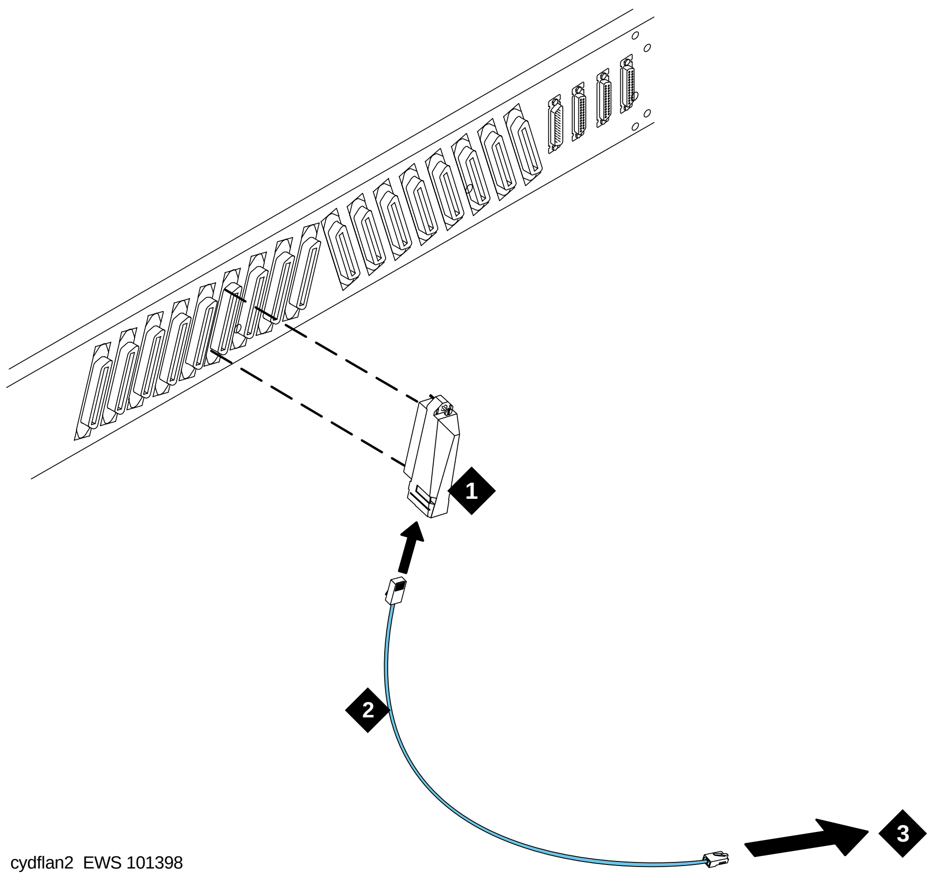


Figure Notes

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

Figure 3. Cable Connection for C-LAN Connectivity

Assign node names

This task assigns node names and IP addresses to each node in the network. You must administer the Node Names screen on each switch in the network.

The node names and IP addresses in any network should be assigned in a logical and consistent manner from the point of view of the whole 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.

NOTE:
Enter node names for Intuity AUDIX and MSA adjuncts on page 1. Enter node names for switches, routers, and CMS starting on page 2.

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

```
change node-names ip                                Page 1 of 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 via the CLAN board.
IP Address	The unique IP addresses of the nodes named in the previous field.

3. Submit the screen.

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.

Alternate Gatekeeper and C-LAN load balancing

Alternate gatekeeper gives IP endpoints a path to any available C-LAN circuit packs.

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 MultiVantage solution uses LAN regions based on IP address, the software also sends the IP addresses of CLANs in interconnected regions. These C-LAN addresses are 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.

If the network connection to one C-LAN circuit pack fails, the IP endpoint re-registers with a different C-LAN.

IP Telephones can be programmed to search for a gatekeeper independent 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 into the DCHP server.

Alternate gatekeeper and registration addresses, and C-LAN circuit pack load-sharing, spread IP endpoint registration across more than one C-LAN circuit pack. This increases system performance and reliability.

C-LAN load balancing and alternate gatekeeper addresses require IP stations that accept multiple IP addresses: IP telephone, IP softphone, Avaya IP Agent or Avaya R300.

Endpoint capabilities

Endpoint	Number of Gatekeepers	How set
IP Telephone	1	default - DNS name AvayaCallServer, or manually, one fixed IP address
	8	via DHCP - DNS names of fixed IP addresses. DHCP limits all options to a total of 128 bytes.
	10	via TFTP - DNS names of fixed IP addresses. TFTP overwrites any gatekeepers provided by DHCP
	30	fixed IP addresses from the Avaya MultiVantage software
IP SoftPhone R3	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 the Avaya MultiVantage software
Avaya R300	2	manually through the Avaya R300 administration

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.

TN2302 IP Media Processor

Use the TN2302 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 TN2302 IP Media Processor board supports the T.38 standard for fax transmissions.

The TN2302 IP Media Processor provides reduction of per-port costs and improved voice quality through its dynamic jitter buffers. The TN2302 IP Media Processor includes a 10/100BaseT Ethernet interface to support H.323 endpoints for IP trunks and H.323 endpoints.

The TN2302 IP Media Processor can perform echo cancellation, silence suppression, DTMF detection, and 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)

Transmission interface

The TN2302 IP Media Processor provides improved voice quality through its dynamic jitter buffers. The TN2302'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. This inserted dB change is under switch software control. System administrators can administer loss/gain, based on country code on the Terminal-parameters screen.

Hairpinning

The TN2302 IP Media Processor supports 64 ports of shallow hairpin. IP packets that do not require speech coder transcoding can be looped back at the UDP/IP layers with a simple change of addressing. This reduces delay and leaves DSP resources available.

Facility test calls

The TN2302 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 TN2302's ports.

Survivable remote EPN

Any survivable remote EPN containing a C-LAN board and H.323 station sets should also contain a TN2302 IP Media Processor.

Universal DS1 (UDS1) Circuit Packs

Universal DS-1 circuit packs have the same functionality as the other DS1 circuit packs, with the addition of echo cancellation circuitry. The UDS1 offers echo cancellation tail lengths of up to 96 milliseconds (ms).

The UDS1 circuit packs 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 MultiVantage 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 a data call. Echo cancellation is also selectable per channel, even though it is administered on a trunk group basis.

Echo Cancellation on UDS1 Circuit Packs

Determine if echo cancellation is enabled for UDS1 circuit packs.

NOTE:
Starting with Release 10, the System Parameters Customer Options screen is display only. 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.

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

Administer Echo Cancellation on Trunks

Echo cancellation on the DS1 circuit pack applies to voice channels. DS1 circuit packs support the following trunk group types:

- CO
- FX
- WATS
- DID
- DIOD
- DMI-BOS
- ISDN-PRI
- TIE

Administer the trunk group

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

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

  Abandoned Call Search? _
  Suppress # Outpulsing? _

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

2. Move to the following field

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

3. Submit the screen.

Administer DS1 Circuit Pack

1. Type **add ds1 <port>** to open the DS1 Circuit Pack screen.

```
add ds1 01c0401                                                  Page 1 of 2
                        DS1 CIRCUIT PACK

             Location: 01C04                     Name: _________
             Bit Rate: _____              Line Coding: ____
    Line Compensation: _                 Framing Mode: ___
       Signaling Mode: isdn-pri__
              Connect: _________            Interface: ___________
    TN-C7 Long Timers?                Country Protocol: ____
 Interworking Message:                Protocol Version: _
            Idle Code: ________                   CRC? _
            Idle Code: ________
                            DCP/Analog Bearer Capability: ________

       Slip Detection? _               Near-end CSU Type: ________
    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.
EC Direction	Indicates the direction of the echo that is being cancelled. Enter inward or outward.
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, 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.

Administer Echo Cancellation per Channel

1. Enter **add personal-co-line** to open the 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.

IP Server Interface (TN2312AP)

The IP Server Interface, (TN2312AP) provides transport of CCMS messages over IP allowing the S8700 Media Server to communicate with the Port Networks. The IP Server Interface (IPSI) is used to wrap/unwrap the CCMS messages from the IP wrapper and place the messages onto the control backplane.

All IPSI boards connect to the AVAYA S8700 Media Server via Ethernet. When this connection is not possible or desirable, the server can also control a port network through one IPSI that is connected to the server via Ethernet and through the bearer network (direct connect, traditional center stage or ATM) to a port network with an expansion interface. One IPSI can support up to 5 Port Networks, including the port network hosting the IPSI, with heavy traffic.

The following are characteristics of the IP Server Interface:

- Always resides in the tone/clock slot
- 10/100BaseT interface for connection to server
- 10/100BaseT interface for connection to services laptop
- Port network clock generation and synchronization (Stratum 4 type II only)
- Port network tone generation
- Port network tone detection/global call classifier/international protocols
- Port network packet interface
- Supports remote firmware download
- Interface to port network TN775D maintenance board
- Serial number support for the License File Feature Activation

The IPSI connects the S8700 Media Server to the PN and carries the control network signals to the other boards in the PN. It provides network packet, tone clock, and tone detection and receiving functionality.

When used in S8700 Multi-Connect configurations, IPSI IP addresses are typically assigned automatically using DHCP service. Also, a dedicated IPSI Ethernet connection to a laptop can be used to assign static IP addresses or for maintenance and firmware downloads. When used in S8700 IP Connect configurations, IPSI IP addresses are always static.

IPSI Installation

For information about how to install and administer the IPSI circuit pack, see the installation document for your system.

IP Networking Administration

3

This chapter contains instructions for the initial administration of H.323 trunks, IP-IP direct connections, hairpinning, IP Telephones, and IP softphones. This chapter also contains a description of IP serviceability tools and troubleshooting of IP networks.

Overview

IP Solutions use the TN2302 IP Media Processor for voice processing and the TN799 C-LAN for signaling.

H.323 Trunks

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 a reduction in long distance voice and fax expenses, facilitation of global communications, full-function networks with data and voice convergence, and network optimization by using the existing network resources.

H.323 Trunk

The TN2302 enables H.323 trunk service using IP connectivity between a Multi-Vantage 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, as generic tie-trunks permitting interconnection with other vendors' H.323 v2-compliant switches, or as direct-inward-dial (DID) type public trunks providing access to the switch for unregistered users.

H.323 Trunk Administration

This section describes the administration steps needed to set up H.323 trunks.

Pre-administration task summary

Before you can set up a H.323 trunk, perform the following tasks:

√	Task
	Verify customer options for H.323 trunking
	Administer C-LAN and IP Media Processor circuit packs. **NOTE:** These circuit packs are not required if your system has built-in Ethernet capabilities.
	Administer QoS parameters by setting thresholds for network performance (see "Quality of Service and voice quality Administration" on page 4-64).
	Administer additional QoS parameters, including defining IP Network Regions and specifying the codec type to be used (see "Quality of Service and voice quality Administration" on page 4-64).
	Assign IP node names and IP addresses to each node in the network.
	Define the IP interface for each C-LAN, MedPro, and IP Media Processor on the switch.
	Administer an Ethernet data module for the connection between the C-LAN board and the LAN.
	Optionally, implement Best Service Routing (BSR).

Verify 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.

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

```
                                                              Page   1 of X
                          OPTIONAL FEATURES
                                                                        Used
           G3 Version:                               Maximum Ports: 100
             Location:                    Maximum XMOBILE Stations: 0
             Platform:

IP PORT CAPACITIES
                              Maximum Administered IP Trunks:
                 Maximum Concurrently Registered IP Stations:
                   Maximum Administered Remote Office Trunks:
      Maximum Concurrently Registered Remote Office Stations:
                    Maximum Concurrently Registered IP eCons:
Maximum Number of DS1 Boards with Echo Cancellation:
                                 Maximum VAL Boards:
```

2. Verify that the following fields have been completed:

Field	Conditions/Comments
G3 Version	This value should reflect the current version.
Maximum Administered IP 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 H.323 trunks field.

```
                                                       Page   3 of 8

                        OPTIONAL FEATURES

    Emergency Access to Attendant?            ISDN Network Call Redirection?
            Enable 'dadmin' Login?                          ISDN-BRI Trunks?
           Extended Cvg/Fwd Admin?                                 ISDN-PRI?
            Enhanced Conferencing?                    Local Spare Processor?
                  Enhanced EC 500?                     Malicious Call Trace?
      External Device Alarm Admin?     Mode Code for Centralized Voice Mail?
                 Flexible Billing?                   Multifrequency Signaling?
    Forced Entry of Account Codes?  Multimedia Appl. Server Interface(MASI)?
       Global Call Classification?         Multimedia Call Handling (Basic)?
              Hospitality (Basic)?      Multimedia Call Handling(Enhanced)?
  Hospitality (G3V3 Enhancements)?                       Multiple Locations?
                       IP Trunks?
            IP Attendant Consoles?
                     IP Stations?          Personal Station Access (PSA?)?
               ISDN Feature Plus?
```

4. Verify that H.323 Trunks and ISDN-PRI are enabled. If not, you need to obtain a new license file.

Administer C-LAN and IP Media Processor circuit packs

Follow these steps to administer the C-LAN and IP Media Processor circuit packs:

1. Type **change circuit-packs** to open the 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 board, complete the following fields:

Fields for CLAN	Conditions/Comments
Code	TN799
Name	C-LAN

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

Fields for IP Media Processor	Conditions/Comments
Code	TN2302
Name	IP Media Processor displays automatically

4. Submit the screen.

Administer QoS parameters

Four parameters on the Maintenance-related System Parameters screen determine threshold 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 "Set thresholds for network performance of IP trunking").

Administer additional QoS parameters

Administer additional QoS parameters, including defining IP Network Regions and specifying the codec type to be used. See Chapter 4, "Quality of Service and Voice Quality Administration".

Assign IP node names

Avaya MultiVantage software 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 that this switch communicates with via IP connections. The Node Names screen must be administered on each switch in an IP network.

A node can be:

- a C-LAN Ethernet or PPP port
- a bridge or router
- a CMS Ethernet port
- or an Intuity AUDIX or other MSA network interface card

Enter the AUDIX and MSA node name and IP address on the Audix and MSA Node Names screen. 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 Node Names, do the following:

1. Type **change node-names ip** to open the 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.

Define IP interfaces

The IP interface for each C-LAN and Media Processor board 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, do the following:

1. Type **change ip-interfaces** to open the IP Interfaces screen.

```
change ip-interfaces                                                Page 1 of 4

              IP INTERFACES

Enable                                                                      Net
Eth Pt  Type   Slot   Code Sfx Node Name       Subnet Mask    Gateway Address Rgn
   y   C-LAN  01A10  TN799 C clan-a1          255.255.0.0     192.168.10 .100 1
   y   MEDPRO 01A13  TN2302   medpro-a1       255.255.0.0     192.168.10 .100 1
   y   MEDPRO 01A16  TN2302   medpro-a2       255.255.0.0     192.168.20 .100 1
   y   C-LAN  01A09  TN799 D clan-a2          255.255.0.0     192.168.20 .100 2
   n                                          255.255.255.0
   n                                          255.255.255.0
   n                                          255.255.255.0
```

2. Complete the following fields as shown:

Field	Conditions/Comments
Enable Eth Pt	Enter y to use the Ethernet port. Enter n to disabled the port before you make changes to its attributes on this screen.
Type	Enter **C-LAN** or **MEDPRO**.
Slot	Enter the slot location for the circuit pack.
Code	Display only. This field is automatically populated with TN799 for C-LAN or TN2302 for IP Media Processor.
Sfx	Display only. This field is automatically populated with the prefix of the circuit pack.
Node name	Enter the node name for the IP interface. This node name must already be administered on the IP Node Names screen.
Subnet Mask	Enter the subnet mask associated with the IP address for this IP interface. The IP address is associated with the node name on the Node Names screen.
Gateway Addr	Enter the address of a network node that serves as the default gateway for the IP interface.
Net Rgn	Enter the region number for the IP interface. Enter a value between 1-44.

3. Submit the screen.

Assign Link via Ethernet Data Module to the LAN

Use this screen to administer an Ethernet data module for the connection between the C-LAN board'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 via 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 via an Ethernet data module, do the following:

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

```
add data-module next                                              Page   1 of 2
                                DATA MODULE

   Data Extension: 2377          Name: C-LAN                          BCC:
             Type: Ethernet       COS: 1      Remote Loop-Around Test?
             Port: 01c0817        COR: 1        Secondary data module?
              ITC:                                         Connected To:

ABBREVIATED DIALING
       List 1:
SPECIAL DIALING OPTION:
ASSIGNED MEMBER:
          Ext:
```

2. Complete the following fields as shown:

Field	Conditions/Comments
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 when the private network contains only Avaya MultiVantage switches and adjuncts. Enter n when the network includes non-Avaya MultiVantage switches that use the 0’s method of forming broadcast addresses.

3. Submit the screen.

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.

H.323 trunk administration

After you have finished the pre-administration tasks, you need to perform the following administration to set up H.323 trunks:

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

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.

Create a 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 that this switch is connected to via 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 MultiVantage Software*.

To create a signaling group, do the following:

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

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

Group Number: 1            Group Type: h.323
                        Remote Office?
                                              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

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
                                       Bypass If IP Threshold Exceeded? n
                                        Direct IP-IP Audio Connections? n
                                                  IP Audio Hairpinning? n
                                        Internetworking Message: PROGress
```

2. Complete the following fields as shown:

Table 3. Signaling Group screen options

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.
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.
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 MultiVantage solution. Enter y when the far-end switch requires a location request to obtain a signaling address in its signaling protocol.

Continued on next page

Table 3. Signaling Group screen options *(Continued)*

Field	Conditions/Comments
Calls Share IP Signaling Connection	Enter y for connections between Avaya MultiVantage solutions. Enter n when the local and/or remote switch is not an Avaya MultiVantage solution.
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.

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                                                     Mach.
 Index    Ext.    Enabled  Established   Dest. Digits    Appl.       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.

Create a trunk group

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.

To create a trunk group, do the following:

1. Type **add trunk-group next** to open the 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 Lucent? n
                                                       Digital Loss Group: 13
Calling Number - Delete:     Insert:                     Numbering 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 an Avaya MultiVantage switch.

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

```
add trunk-group next                                            Page   2 of  10
                                TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                               Data Restriction? n       Maintenance Tests? y
                                      Send Name: n     NCA-TSC Trunk Member:
                                                        Send Calling Number: n
            Used for DCS? y       PBX ID:
   Suppress # Outpulsing? n                                 DCS Signaling: d-chan
 Outgoing Channel ID Encoding: exclusive     UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send Connected Number: n
             Send UUI IE?
               Send UCID? n                           DS1 Echo Cancellation?
 Send Codeset 6/7 LAI IE? y                USNI Delayed Calling Name Update?
                            Network (Japan) Needs Connect Before Disconnect?
```

4. Complete the following fields as shown:

Field	Conditions/Comments
Used for DCS	Enter y.
PBX ID	Enter the switch ID.
DCS Signaling	Enter d-chan.
Send Name Send Calling Number Send Connected Number	These fields may need to be changed if the far-end is a not an Avaya MultiVantage solution.

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

```
add trunk-group next                            Page   4 of  10
                                TRUNK GROUP
                           Administered Members (min/max):   0/0
GROUP MEMBER ASSIGNMENTS       Total Administered Members:   0

       Port    Code Sfx Name        Night           Sig Grp
  1: ip        H.323 Tr 1                             3
  2: ip        H.323 Tr 2                             3
  3: ip        H.323 Tr 3                             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.
Sig Grp	Enter the number of the signaling group that you set up for H.323 trunks.

Modify the signaling group

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

To modify a 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.

```
                                                                 Page 1 of 5
                                  SIGNALING GROUP

Group Number  ___            Group Type: h.323
                         Remote Office?
                                              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

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
                                       Bypass If IP Threshold Exceeded? n
                                        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 will be the one that accepts incoming calls.

4. Submit the screen.
5. Type **release signaling-group <number>** to release the signaling group.

IP SoftPhones

IP SoftPhones operate on a PC equipped with Microsoft Windows and with TCP/IP connectivity to an Avaya MultiVantage solution. 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 consists of a PC running the Avaya IP SoftPhone application and Avaya iClarity IP Audio, with a single IP connection to an Avaya MultiVantage server.
- Telecommuter 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.
- The Native H.323 mode is a PC-based single phone with limited features. Avaya iClarity IP Audio is not supported with in Native H.323 mode.

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 the Avaya MultiVantage software to support IP SoftPhones are given in *Administrator's Guide for Avaya MultiVantage Software.*

IP SoftPhone administration

This book 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 covered in *Administrator's Guide for Avaya MultiVantage Software*.

There are two main types of IP SoftPhone configurations: the telecommuter and the road-warrior.

The MultiVantage software 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.

The telecommuter uses two connections to the Avaya MultiVantage solution: a connection to the PC over the IP network and a 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.

Administering a phone in telecommuter mode

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.

1. Verify that IP SoftPhone is enabled.

   Type display system-parameters customer-options.

   Review the following fields on the System-Parameters Customer Options 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.
Maximum Concurrently Registered IP Stations	Specifies the maximum number of IP stations that are simulated registered, not the maximum number that are simultaneously administered. This value must be greater than 0.
IP Stations	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, 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. Go to 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; Service Link Mode: **as-needed**
4. Install the IP SoftPhone software on the user's PC.

Road-warrior

The road-warrior uses two separate software applications running on a PC that is connected to an Avaya MultiVantage solution 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 R2 or greater, iClarity is automatically installed to handle voice communications.

Administering a phone in road-warrior mode

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.

1. Verify that IP SoftPhone is enabled.

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

   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**.
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** for every product except the Avaya R300. For the Avaya R300, 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. 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. Go to 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.
Security Code	Enter the user’s password.
IP SoftPhone	Enter **y**.

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

5. Install the IP SoftPhone software on the user's PC (iClarity automatically installed with the IP SoftPhone R2 or greater).
6. For pre-R2 IP SoftPhones, an H.323 V2-compliant audio application (such as Microsoft NetMeeting) must be installed.

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.

46xx IP phone series

The 46xx IP Telephone product line possesses a number of shared model features and capabilities. All models also have downloadable firmware, automatic IP address resolution via DHCP, and manual IP address programming.

Some features and capabilities are distinctive to each of the IP Telephone models.

4602 IP telephone

The 4602 has two call appearance buttons, a drop button and a voice mail retrieval button that provides automatic access to voice mail.

4606 IP telephone

The 4606 has six line appearance/feature button assignments, a full-duplex speakerphone, an infrared Ethernet interface, and an Ethernet repeater. The 4606 does not support softkey button assignments.

4612 IP telephone

The 4612 has twelve line appearance/feature button assignments, a full-duplex speakerphone, an infrared Ethernet interface, and an Ethernet repeater. The 30A Ethernet switch is available as an optional stand. The 4612 supports twelve softkey button assignments.

4620 IP screenphone

The 4620 IP Screenphone has a large gray-scale screen with six buttons on either side for feature access, a full-duplex speakerphone, an infrared Ethernet interface, and an Ethernet switch. It has optional feature expansion module and downloadable call appearance/ feature button information. The 4620 IP phone does not need paper labels, as the button information appears on a screen on the phone.

4624 IP telephone

The 4624 has 24 line appearance/feature button assignments, a full-duplex speakerphone, an infrared Ethernet interface, and an Ethernet repeater. The 30A Ethernet switch is available as an optional stand. The 4624 also supports twelve softkey button assignments.

4630 IP screenphone

The 4630 IP Screenphone is referred to as an IP appliance, an endpoint that uses the capabilities of Internet Protocol to provide much more functionality than a basic telephone. The 4630 uses a 1/4 VGA color, touch-sensitive screen for web-enabled applications, as well as the full suite of Avaya MultiVantage telephony features, LDAP directory, and www.messenger voice mail features. The 4630 IP Screenphone has an Ethernet repeater, and the 30A Ethernet switch is available as an optional stand.

Avaya Softconsole IP Attendant

The Avaya Softconsole is a Windows-based application that can replace the physical 302-series console. Attendants use a PC interface to answer and route calls via IP. The Avaya Softconsole supports an IP mode. In this mode, it works with a traditional DCP telephone connected via serial port to an external adapter. In this mode, it can function in either a telecommuter or road warrior configuration.

Installing IP telephones

MultiVantage 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.

Audio capability for the IP Telephones requires the TN2302 IP Media Processor circuit pack, which also provides hairpinning and IP-IP direct connections. This conserves TDM board and timeslot resources and improves voice quality.

The 46xx IP Telephone also requires a TN799 Control-LAN (C-LAN) circuit pack for the signaling capability on the CSI, SI, and R platforms. You do not need a C-LAN board to connect an IP Telephone if your system has built-in Ethernet capability.

To install an IP Telephone:

Install Circuit Packs

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. It is not necessary to power down the cabinet to install the circuit packs.

Software requirements

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

The software treats the 46xx IP Telephones as any new station type to include the capability to list station, display station, change station, duplicate station, alias station, and remove station.

Administering Avaya IP telephones

IP Telephones R1.5 or greater use a single connection, and you only need to administer the station type. To use IP Telephones, the IP and H.323 station customer options need to be enabled in the license file.

To add an IP telephone:

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

```
add station next                                                 Page 1 of 5
                                 STATION

       Extension:                Lock Messages? n                    BCC: 0
            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 Station? 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.
Port	Enter **x**.

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 (R1.0), 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.

IP-IP Direct Connections, Hairpinning, Inter-Network Management Features

This section describes IP-IP direct connections (shuffling), hairpinning, and inter-network region connection management. IP-IP direct connections and hairpinning help reduce per-port costs by freeing up Digital Signal Processor (DSP) and time slot resources. IP-IP direct connections and hairpinning also help reduce IP bandwidth usage.

Hairpinning and IP-IP direct connections reduce the number of codec conversions that a voice call encounters, which helps improve the voice quality users experience over IP calls. IP-IP direct connections and shallow hairpinning also reduce the number of media conversion ports required.

IP-IP direct connections reroutes the voice channel between two IP endpoints in one of two ways. It either reroutes an IP-TDM-IP voice channel to a direct IP-IP connection, or it reroutes a direct IP-IP voice channel to an IP-TDM-IP voice channel.

Hairpinning is described as: rerouting the voice channel connecting two IP endpoints so that the voice goes through the TN2302 Media Processor board in IP format, without having to go through the TDM bus. Only the IP and Real-Time Protocol (RTP) packet headers are changed as the packet goes through the TN2302 Media Processor Board. This requires that both endpoints be using the same codec.

A TN2302 Media Processor board is required in order to set up an IP-media processor-IP hairpin connection. Both the TN2302 Media Processor, and the Medpro circuit packs are capable of accepting the commands to shuffle an audio connection. Almost any IP endpoint can be an endpoint on a hairpinned connection, although system administrators may choose to restrict hairpin connections per switch, region, signaling group, or to certain endpoints.

For most user actions, the switch will behave the same way for an IP phone as it does for any other Digital Communications Protocol (DCP) set. Similarly, from a user's point of view, for most user actions, the switch will behave the same way for calls carried over a LAN by IP-IP direct audio connections and IP-media processor-IP hairpinned connections as it does for calls switched through the TDM bus.

Detailed description

IP-IP direct connections and hairpinning have the following subfeatures:

- Audio hairpin connections
- Audio IP-IP direct connections
- Inter-network region connection management

Audio hairpin connections

Avaya MultiVantage software provides audio IP-media processor-IP hairpinning on the Media Processor (TN2302 circuit pack) to conserve resources on the board and time slots on the Time Division Multiplexed (TDM) bus and to improve voice quality by eliminating use of codecs on the media processor. The end user does not have to take any action to invoke audio hairpinning. The switch will hairpin, if necessary, after a point-to-point call is established between two voice endpoints.

The media processor typically accomplishes audio hairpinning within approximately 50ms intervals (converting a TDM based connection to/from shallow hairpin connection). The media processor preserves media stream integrity such as Real-Time Protocol (RTP) headers.

Audio IP-IP direct connections

Avaya MultiVantage software provides IP-IP direct audio connections to save switch resources, such as channels on the media processor and TDM bus time slots, and to improve voice quality by eliminating extra coder/decoder functions on the media processor. This is sometimes called “shuffling.” The end user does not have to take any action to invoke audio shuffling. The switch will shuffle, if necessary, after a point-to-point call is established between two voice endpoints, provided both endpoints are capable of handling shuffling.

If the software determines that shuffling is not supported on both legs of the call, it may attempt to hairpin the audio connection. The software may attempt to shuffle IP audio connections as soon as the point-to-point connection between two endpoints is established. If the endpoints take some actions that require the switch to insert a tone plant in the connection, the audio connection will be converged back to the TDM bus through the media processor.

NOTE:
Avaya MultiVantage cannot shuffle a call that is connected to the switch vial Network Address Translation (NAT).

Inter-network region connection management

For calls requiring direct IP-IP media connection between two regions that are interconnected, the software checks for path availability and also ensures that the path meets the performance criteria determined by the system administrator. The switch performs a periodic background ping test between each interconnected region to determine the status of the connections.

You can establish a roundtrip propagation delay threshold for calls over IP trunks. If the threshold is exceeded, the calls will reroute over other available trunks, until the performance of the IP trunks improves.

Feature limitations

The TN2302 Media Processor board provides up to 64 audio channels. They may be either:

- channels of hairpin IP-IP. Each hairpin call needs two channels per board, one to each endpoint.
- channels of G.711 IP-TDM
- channels of G.729 or G.723-TDM. Each G.729 or G.723 call needs two channels per board.
- channels of Fax IP-TDM. Each Fax IP-TDM call needs two channels per board.

Hairpin and IP-IP direct calls continue to use sockets on a C-LAN board or on a server with embedded Ethernet capability. All three types of connections; IP-TDM, IP-IP hairpin, and IP-IP direct, still need call-control sockets. When registered, a single-connect set uses one socket, a dual-connect set in “permanent” mode uses two sockets, and a dual-connect set in “as-needed” mode uses one socket plus one additional socket whenever active on a call.

Audio hairpin connection setup

The following conditions must be met to connect audio endpoints with an IP-media processor-IP hairpin:

- A point-to-point voice connection must exist between two endpoints.
- The endpoints must be in the same LAN region or in interconnected LAN regions.
- A single media processor (TN2302 Media Processor) must serve both endpoints.
- The endpoints must use a single common codec. Two different speeds of G.723 count as one codec.
- The endpoints must be administered to allow hairpinning.
- Everything in hairpinning is satisfied and shuffling is not allowed.

When a multiparty call is reduced to a two-party call (via drop or hang-up), the resulting connection is hairpinned only if the remaining two parties satisfy the conditions or the requirement for Hairpinned Setup.

Audio shuffling setup

The following conditions must be met to connect the audio of two IP endpoints in an IP-IP direct connection:

- Both IP endpoints must be administered to allow shuffling.
- A point-to-point voice connection must exist between two endpoints and no active call (in-use or held) exists on either endpoint that requires TDM connectivity (such as applying tones, announcement, or conferencing).
- The endpoints must be in the same LAN region or in interconnected LAN regions.
- The inter-region connection management rules are must be met.
- There is at least one codec in common between the codec lists of the endpoints involved and the Inter-network region Connection Management codec list.
- The endpoints must have at least one codec in common as shown in their current codec negotiations between the endpoint and the switch.

Inter-network region connection management

The Roundtrip Propagation Delay threshold is administered on the System-Parameters IP-options screen. In order for the software to connect two network regions, the connection must meet the Roundtrip Propagation Delay threshold, and the network regions must be administered on the Network Region screen.

System resets

On a level 2 or greater restart, all IP network endpoints are cleared, calls are dropped, and endpoints are unregistered. Hairpinned calls and TDM calls are dropped.

What happens to calls carried by IP-IP direct audio connections on a level 2 or greater restart depends on the endpoints involved in the call. On a level 2 or greater restart, the set on such calls will lose registration. Avaya IP telephones do not drop in-progress calls when they lose their registration.

Transmission interface

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 via a TN2302 Media Processor, the board will adjust the levels to 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.

Adjust loss to USA DCP levels

The switch will instruct the TN2302 Media Processor board 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.

Voice levels are not administrable on shuffled calls

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

Preference Order

Note that the loss to be applied during a call carried by an IP-IP direct or an IP-media processor-IP hairpin audio connection is constant during shuffling or hairpinning of all three types of connections:

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

Interactions

If sufficient audio channels are not available when an IP direct (shuffled) or IP hairpinned call needs to return to the TDM bus, the switch blocks the user’s action and notifies the user via a lamp flutter. There are several features that require calls to return to the TDM bus.

Adjunct Switch Applications Interface (ASAI)

The switch delays sending an acknowledgement to the ASAI adjunct that the ASAI instruction has been completed until after all necessary resources to set up the audio channels between the sets and the TN2302 Media Processor's are allocated if the following are true:

- The two endpoints' audio paths are directly IP-IP connected together, and
- ASAI issues an instruction that causes the switch to reconnect the speech paths back to a TN2302 Media Processor port,

The following ASAI instructions may cause the switch to reconnect the speech paths back to a TN2302 Media Processor port:

- Selective listening
- Third Party make call
- Third Party take control
- Third Party selective hold
- Third Party Single Step Conference
- Third Party merge
- Third Party Selective Drop
- Third Party Clear Call
- Third Party send DTMF
- Third Party Answer
- Third Party Auto Dial
- Request Feature.

Whether or not these instructions cause the switch to return the call to IP-TDN-IP will depends on other options, such as whether the switch needs to supply music on hold, for example.

If there are insufficient ports available on TN2302 Media Processor to carry out the above commands, the switch will send an appropriate Denial (NAK) Cause value back to the ASAI adjunct.

Attendant Console

If a station is on an IP-media processor-IP hairpin call or a call carried over an IP-IP direct audio connection, that station will still show up as busy on the attendant busy lamp field and on Busy Indication buttons.

Bridging

Circuit-switched endpoint bridged with IP endpoints

If endpoint A and endpoint B have an audio connection to each other and user C has a bridged call appearance of endpoint A on C's set, the connection between A and B can still be made via IP-media processor-IP hairpin, or IP-IP directly. User C is not considered a third party to this call for hairpin setup or audio shuffling setup, unless user C selects the bridged call appearance.

Bridging and new connections

The switch will not set up a direct IP-IP or and IP-media processor-IP connection between two endpoints while either endpoint is bridged in a call with additional parties.

Hold

If user A, user B, and user C are talking on a conference call, and user C presses the hold button, leaving A and B talking together, the switch will not set up a direct IP-IP or IP-media processor-IP hairpin connection between those two endpoints as long as C keeps them on hold. This prevents a delay when C re-enters the call.

Music On Hold

Music On Hold uses network and TN2302 circuit pack resources.

Soft Hold

The switch will not set up an IP-IP direct or an IP-media processor-IP hairpin connection between two endpoints while either endpoint has a TDM party on soft hold awaiting a transfer.

Manual Signaling

If endpoint A and endpoint B have an audio connection to each other via IP-media processor-IP hairpinning or IP-IP directly, and a third endpoint C uses manual signaling to ring endpoint A, A and B will remain IP-media processor-IP hairpinned or IP-IP directly connected together.

Shuffling and hairpinning administration

For shuffling and hairpinning to work, it must be administered properly for the endpoint, the signaling group, and the network region. The system-wide parameters must also be established. I

To administer shuffling and hairpinning, the H.323, IP Stations and/or Remote Office fields must be enabled on the Customer Options screen. If not, you need to obtain a new license file.

Administering system parameters for hairpinning and shuffling

To administer the system to allow hairpinning and shuffling:

1. Type **change system-parameters features** to open the Feature-Related System Parameters screen.

```
change system-parameters features                                 Page 12 of 12
                FEATURE-RELATED SYSTEM PARAMETERS

AUTOMATIC EXCLUSIONS PARAMETERS

                            Automatic Exclusion by COS? n

                                   Recall Rotary Digit: 2
                         Password to Change COR by FAC:
              Duration of Call Timer Display (seconds):
WIRELESS PARAMETERS
Radio Controllers with Download Server Permission (enter board location)
                        1:     2:     3:      4:     5:
IP PARAMETERS
                        Direct IP-IP Audio Connections? y
                                  IP Audio Hairpinning? y
RUSSIAN MULTI-FREQUENCY PACKET SIGNALING
                                                Re-try? n
            T2(Backward Signal)Activation Timer (secs):20
```

2. Administer the following fields:

Field	Value
Direct IP-IP Audio Connections	y
IP Audio Hairpinning	y

3. Submit the screen.

Administering a station for hairpinning and shuffling

You administer shuffling and hairpinning on a per-station basis. The specific station types that can be administered are the 46xx series IP phones, Avaya Softconsole, H.323 stations, and the station types that can be administered as IP softphones, for example those with the IP softphone field on the station screen.

To administer shuffling or hairpinning for an IP station:

1. Type **change station xxx** to open the Station screen.

```
change station 12345                                        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? y Per Station CPN - Send Calling Number?
         Service Link Mode: as-needed
           Multimedia Mode: enhanced    Audible Message Waiting?
      MWI Served User Type:           Display Client Redirection? n
                                     Select Last Used Appearance? n
                                       Coverage After Forwarding? n
           Automatic Moves:              Multimedia Early Answer? n
    Emergency Location Ext:      Direct IP_IP Audio Connections? n
                                            IP Audio Hairpinning? n
```

2. Administer the following fields:

Field	Value
Direct IP-IP Audio Connections	y
IP Audio Hairpinning	y

3. Submit the screen.

Administering a signaling group for hairpinning and shuffling

You administer shuffling and hairpinning for trunks in the H.323 signaling group. To administer a signaling group for shuffling and hairpinning:

1. Type **change signaling group xxx** to open the 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
                                                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? n

  Near-end Node Name: mipsn01A        Far-end Node Name: dr99
Near-end Listen Port: 1800          Far-end Listen Port: 1800
                                 Far-end Network Region:
   LRQ Required? n                 Calls Share IP Signaling Connection? n
   RRQ Required? n
                                       Bypass If IP Threshold Exceeded? n
                                        Direct IP-IP Audio Connections? y
                                                  IP Audio Hairpinning? y
                                        Internetworking Message: PROGress
```

2. Administer the following fields:

Field	Value
Direct IP-IP Audio Connections	y
IP Audio Hairpinning	y

3. Submit the screen.

Administering a network region for hairpinning and shuffling

You must administer shuffling and hairpinning for a given network region in order for endpoints within that region to use these capabilities. If a call is moving to direct IP-IP, the software checks each endpoint's region to ensure that the move is allowed. If a call is moving to IP-media processor-IP, the software checks each endpoint's region, as well as the region of the TN2302 Media Processor.

To administer the network region to allow hairpinning and shuffling:

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

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

      Region: 1
        Name:

Audio Parameters                 Direct IP-IP Audio Connections? y
   Codec Set: 1                            IP Audio Hairpinning? y
    Location:
  UDP Port Range                                   RTCP Enabled? y
         Min: 2048               RTCP Monitor Server Parameters
         Max: 65535               Use Default Server Parameters? n
                                             Server IP Address: ___.___.___.___
DiffServ/TOS Parameters                            Server Port: _____
 Call Control PHB Value: 46           RTCP Report Period (secs): __
   VoIP Media PHB Value: 46
           BBE PHB Value:             Resource Reservation Parameters
                                                      RSVP Enabled? y
             802.1p/Q Enabled? y            RSVP Refresh Rate (secs):
 Call Control 802.1p Priority: 6     Retry upon RSVP Failure Enabled? y
   VoIP Media 802.1p Priority: 6                 RSVP Profile: guaranteed-service
                  802.1Q VLAN: 0
```

2. Administer the following fields:

Field	Value
Direct IP-IP Audio Connections	y
IP Audio Hairpinning	y

3. If C-LAN and IP Media Processor resources are shared between network regions, go to the Inter Network Region Connection Management page of the Network Region screen.

NOTE:
You will not be able to connect IP endpoints in different network regions or share resources between or among network regions unless you make an entry in this page specifying the CODEC set to be used.

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

Region                   (Group of 32)
          1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2
001-032   1 _ _ 2 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
033-064   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
065-096   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
097-128   _ _ 5 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
129-160   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
161-192   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
193-224   6 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
225-256   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
257-288   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
289-320   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
321-352   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
353-384   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
385-416   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
417-448   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
449-480   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
481-500   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
```

4. Specify CODEC sets for your shared network regions as done in the example above. In the example, network region 1 shares resources with the following other network regions, using the specified CODEC sets:
   - network region 1 using CODEC set 1.
   - network region 4 using CODEC set 2.
   - network region 99 using CODEC set 5.
   - network region 193 using CODEC set 6.
5. Submit the screen.

Quality of Service and Voice Quality Administration

4

This chapter provides instructions for administering the Avaya MultiVantage software using Quality of Service (QoS) techniques.

Quality of Service Overview

Quality of Service (QoS) or QoS techniques are a set of policies that improve the behavior of traffic through the IP network. These techniques help you achieve the best possible audio quality when voice traffic is carried over packet facilities.

QoS techniques are applied in two phases:

1. Voice packets are tagged with a priority value through the Avaya MultiVantage Software, or through the data-network.
2. Using that priority tag, the data-network equipment makes decisions about queueing and discarding packets.

Priority policies should be established on your data-network first, and then administered on your system to match.

You can use the following techniques for implementing QoS:

- configurable UDP port ranges
- voice traffic classification using DiffServ and IEEE 802.1p/Q for each network region. Values are set for the IP Media Processor and the TN799 CLAN in separate fields.

You can also improve IP voice quality by utilizing the following features:

- audio CODEC settings for each network region
- multiple CODEC sets with frames-per-packet settings and silence suppression for each CODEC
- network region by signaling group administration (permits different CODEC sets for trunk calls verses station-to-station calls)
- dynamically-adjusted jitter buffers
- hybrid echo cancellation
- packet loss concealment

Avaya Policy Manager (APM), an Avaya product used for QoS administration, also helps define network utilization policies.

Networking factors

There are some key factors of IP networks which determine the quality of voice calls, and therefore drive your QoS policies. These factors include packet delay, jitter, packet loss, bandwidth, transcoding, echo, and silence suppression.

Packet delay

Packet delay is the length of time it takes for a packet to traverse the network. Best practices suggest that the end-to-end one-way packet delay should not exceed 150 ms. Each element of the network adds to packet delay, including switches, routers, distance travelled through the network, firewalls, and jitter buffers. Packet delay exceeding 150 ms may begin to have a noticeable effect; however, some applications or users may elect to tolerate it. A network assessment from Avaya, measuring latency and making recommendations to solve latency issues, is recommended before implementing VoIP solutions.

Jitter

Jitter is a measure of the variance in the time it takes for communication to reach the receiver from the sender, from the application’s perspective. Jitter is generally considered the statistical average variance in delivery time between packets or datagrams.

If this variance exceeds 20 ms it can create voice quality problems. Symptoms of excessive jitter are similar to symptoms of excessive delay, because in both cases packets are discarded.

To compensate for network jitter, many vendors of H.323 voice applications implement a jitter buffer. The purpose of the jitter buffer is to hold incoming packets for a specified period of time before forwarding them, thereby smoothing packet flow.

Packets are discarded if they arrive later than the jitter buffer can hold them. If a packet arrives out-of-order, it is generally due to individual packets being sent across different paths over the network. A packet that arrives out-of-order is generally discarded if they arrive outside the jitter buffer window.

Because of the added packet delay, jitter buffers should be sized to no more than twice the largest statistical variance between packets. The best solution is to have dynamic jitter buffers. These reduce their size as allowed by the network, thereby reducing the delay caused by jitter buffers. Avaya MultiVantage solutions, Avaya IP softphones, and Avaya IP telephones incorporate dynamic jitter buffers.

Packet loss

Network packet loss is when a packet is sent but not received at its final destination. A good quality VoIP network should have packet loss of less than 0.2% between endpoints. Avaya implements packet loss concealment to lessen its impact.

NOTE:
Too much packet delay can be perceived as dropped packets. To a tool which measures packet loss, these dropped packets may appear as lost packets, when in fact they were discarded intentionally.

Bandwidth Control

Bandwidth requirements are a concern to all IP applications, especially those which connect through the WAN. The amount of bandwidth required for an IP connection can be controlled through CODEC selection. CODEC selection requires administration of both the IP Network Region form and the IP Codec Set form (see "Quality of Service and voice quality Administration" in this chapter).

The main difference between CODECs is in the compression algorithm used; Some CODECs compress the voice data more than others. A greater degree of compression results in lower bandwidth requirements on the network, but also introduces transmission delays and lower voice quality.

Depending upon bandwidth availability and required voice quality, it may be worthwhile to select a CODEC that produces compressed audio.

- A G.711 CODEC produces audio uncompressed to 64 kbps. This is the default CODEC, and is recommended for all IP connections within the LAN.
- A G.729 CODEC produces audio compressed to 8 kbps. This is the Avaya recommended CODEC for WAN IP applications.
- G.723.1 CODEC is also supported, but not recommended by Avaya.

If the far-end is a non-Avaya system, you may need to change the CODEC to match one that is supported.

The following table compares CODECs by listing several voice quality considerations. Mean Opinion Score (MOS) is a long-standing subjective method of measuring voice quality. As a guide, you should know that toll-quality voice must achieve a MOS of 4 or above.

Table 4. Comparison of Speech Coding Standards

Standard	Bit Rate (kbps)	MOS
G.711	64	4.3
G.729	8	4.0
G.723.1	6.3 5.3	3.8

Transcoding

When IP endpoints are connected through more than one network region, it is important that each region use the same CODEC. If different CODECs are used by the network regions, delays will occur in packet transmission. The IP Media Processor will be required to act as a gateway translating the different codecs, and an IP-direct (shuffled) connection will not be possible.

Echo

Echo is hearing your own voice reflected back, which happens when a signal on a transmit path appears on a receive path. Echo stems from electrical (imbalanced impedance, cross-talk) or acoustical (room size or speakerphone) sources, and is troublesome if it impairs the ability to have a normal conversation. The difference in time from when a voice enters the network and when it is returned to the originator is known as echo path delay.

Echo cancellers are used to minimize echo. They work by comparing incoming voice patterns to current voice patterns, and cancelling echo when the patterns match. The echo cancellers are limited in their effectiveness by the amount of memory they have, and therefore the amount of voice patterns which they can hold at one time. VoIP can exacerbate the problem when traffic delay exceeds the echo canceller’s memory. Avaya’s echo cancellers are designed for use with VoIP to improve voice quality.

Silence suppression

Voice Activity Detection (VAD) monitors the received signal for voice activity. When no activity is detected for the configured period of time, the software informs the packet voice protocol. This prevents the signal from being sent across the network when there is silence, which saves bandwidth. VADs can greatly conserve bandwidth, but overly aggressive VADs can cause voice clipping and reduce voice quality. When planning for network bandwidth use, especially across WANs, silence suppression is an important factor to consider.

MultiVantage features to support QoS policies

Avaya MultiVantage Software allows you to apply QoS policies across multiple OSI model layers. On the IP Network Region screen there are settings for 802.1p/Q (data link layer, layer 2), Virtual LANs (data link layer, layer 2), DSCP or DiffServ (network layer, layer 3), UDP port ranges (network layer, layer 3), and RSVP (network and transport layers, layers 3 & 4).

802.1p/Q

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 your MultiVantage Software must match your data network.

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. 802.1p is an Ethernet tagging mechanism that can instruct Ethernet switches to give priority to audio frames.

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

802.1Q/p provides for 8 levels of priority and also 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 at the MAC layer.

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 would require a separate network region for each network interface.

Using VLANs

VLANs provide security and create smaller broadcast domains by using software to create virtually separated subnets. Broadcasts are common in most data networks from protocols used by PCs, servers, switches and routers. Creating separate VLANs results in more effective bandwidth utilization and reduces the processor burden on most endpoints.VLANs are defined in data switches, and separate VLANs for voice are strongly recommended.

Differentiated Services (DSCP)

The Differentiated Services Code Point (DSCP or DiffServ) prioritization scheme redefines the Type of Service (TOS) byte in the IPv4 header by combining the first six bits into 64 possible Per Hop Behaviors (PHBs). Set these bits according to your company's policy.

A DiffServ policy must be established across entire IP network, and the DiffServ values used by your MultiVantage Software and by your IP Endpoints must follow this policy.

Configuring ports using UDP

Avaya MultiVantage Software allows users to configure User Datagram Protocol (UDP) port ranges used by VoIP packets. These port ranges are used by data network equipment to assign priority throughout the network.

Integrated Services (RSVP)

RSVP is used to ensure that new calls will not overload a WAN link and disrupt existing calls.

Quality of Service and voice quality Administration

This section contains instructions for administering CODEC sets, defining network regions, and setting network performance thresholds.

Administer IP Codec Set

The IP Codec Set screen allows you to specify the type of CODEC used for voice encoding and companding, compression/decompression.

The IP Codec Set screen

The order in which the CODECs are listed on the following screen is the order of preference of usage. 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 need not be the same, but the order of the listed CODECs must be the same.

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 for each CODEC in the set, based on the number of frames you administer per packet.

To administer an IP Codec Set:

1. Type **change ip-codec-set <#>** to open the IP Codec Set screen.

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

2. Complete the following fields:

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 size 20 ms packets, the following bandwidth is required:

- G.711 A-law - 90 Kbps
- G.711 µ-law - 90 Kbps (used in U.S. and Japan)
- G.723.1-5.3 - 21.3 Kbps
- G.723.1-6.3 - 22.3 Kbps
- G.729 - 34.4 Kbps
- G.729B - 34.4 Kbps

Table 5. IP Codec Set screen fields

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**
Suppress Silence	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)
Packet Size (ms)	Automatically appears.

3. Submit the screen.
4. Type **list ip-codec-set** to list all your CODEC sets on the CODEC set 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
```

5. Review your CODEC sets.

Define IP network region

NOTE:
Avaya strongly recommends that you leave the default values for the following screen.

The IP Network Region screen allows you to define a network region, and to specify which audio transport parameters will be used for communication with endpoints within this region. Audio transport parameters that you can specify on this screen include a CODEC set, Call Control PHB, VoIP Media PHB, Call Control 802.1p priority, VoIP Media 802.1p priority, 802.1Q VLAN ID, and UDP port range.

NOTE:
Although they are not defined on this screen, the IP network region also encompasses a series of resources, media processor boards, that are shared by IP endpoints when they are making calls. Network regions also encompass a series of CLAN circuit packs that are shared by the Load Balancing and Alternate Gatekeeper features. See "Alternate Gatekeeper and C-LAN load balancing" for more information.

The IP-network-region screen

To define an IP network region, do the following:

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

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

      Region: 1
        Name:

Audio Parameters                  Direct IP-IP Audio Connections? y
   Codec Set: 1                             IP Audio Hairpinning? y

    UDP Port Range                                  RTCP Enabled? y
         Min: 2048                RTCP Monitor Server Parameters
         Max: 65535                Use Default Server Parameters? n
                                            Server IP Address: ___.___.___.___
DiffServ/TOS Parameters                           Server Port: _____
 Call Control PHB Value: 46               RTCP Report Period (secs): __
   VoIP Media PHB Value: 46
          BBE PHB Value:               Resource Reservation Parameters
                                                       RSVP Enabled? y
             802.1p/Q Enabled? y            RSVP Refresh Rate (secs):
 Call Control 802.1p Priority: 6     Retry upon RSVP Failure Enabled? y
   VoIP Media 802.1p Priority: 6                RSVP Profile: guaranteed-service
                  802.1Q VLAN: 0
```

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

Field	Conditions/Comments
Name	Describes the region. Enter a character string up to 20 characters.
Codec Set	Specifies the CODEC set assigned to a region. Enter a value between **1-7**. **NOTE:** CODEC sets are administered on the CODEC set screen (see "Administer IP Codec Set").
UDP Port Range-Min	Specifies the lowest port number to be used for audio packets. Enter a value between **2-65406**. **NOTE:** Note: This number must be twice the number of calls 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 Range-Max	Specifies the highest port number to be used for audio packets. Enter a value between **130-65535**. **CAUTION:** *Avoid the range of well-known or IETF-assigned ports. Do not use ports below 1024.*
Call Control PHB Value	Contains the decimal equivalent of the Call Control PHB value. Enter a value between **0-63.** Use PHB **46** for expediting forwarding of packets. Use PHB **40** for audio for legacy systems that only support IPv4 Type-of-Service. This correlates to the older TOS critical setting.
VoIP Media PHB Value	Contains the decimal equivalent of the VoIP Media PHB value. Enter a value between **0-63.** Use PHB **46** for expediting forwarding of packets. Use PHB **40** for audio for legacy systems that only support IPv4 Type-of-Service. This correlates to the older TOS critical setting.
802.1p/Q Enabled	Specifies whether 802.1p MAC-layer prioritization and 802.1Q Virtual LAN specification are enabled for edge devices in this region.

Continued on next page

Field	Conditions/Comments
Call Control 802.1p Priority	Specifies the 802.1p priority value, and appears only if 802.1p/Q field is **y**. Enter **5** or **6** (high). **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 your MultiVantage Software must match your data network.*
VoIP Media 802.1p Priority	Specifies the 802.1p priority value, and appears only if 802.1p/Q field is **y**. Enter **5** or **6** (high). **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 your MultiVantage Software must match your data network.*
802.1Q VLAN	Specifies the 802.1Q virtual LAN value and appears only if the 802.1p/Q field is **y**. **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 your MultiVantage Software must match your data network.*

3. If C-LAN and IP Media Processor resources are shared between regions, go to the Inter Network Region Connection Management screen.

NOTE:
You will not be able to connect IP endpoints in different network regions or communicate between or among network regions unless you make an entry in this page specifying the CODEC set to be used.

```
change IP-network-region 1                                    Page   2 of 2
                 Inter Network Region Connection Management

Region                          (Group of 32)
          1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2
001-032   1 _ _ 2 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
033-064   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
065-096   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
097-128   _ _ 5 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
129-160   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
161-192   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
193-224   6 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
225-256   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
257-288   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
289-320   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
321-352   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
353-384   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
```

4. Specify CODEC sets for your shared network regions as done in the example above. In the example, network region 1 will communicate with the following other network regions, using the specified CODEC sets:
   - network region 1 using CODEC set 1
   - network region 4 using CODEC set 2
   - network region 99 using CODEC set 5
   - network region 193 using CODEC set 6
5. Submit the screen.
6. Type **list ip-network-region qos** to list all your network regions on the IP Network Region 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     34     43    6      7   guaranteed-service 15
 2     Cheyenne           46     34     43    6      7   controlled-load    15
```

7. Type **list ip-network-region monitor** for more information about your network regions on the IP Network Region screen.

```
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   65535
 2     Cheyenne          123.123.123.123  5005      5      1     2048   65535
```

8. Review the network regions with the new audio transport parameters.

Set thresholds for network performance of IP trunking

NOTE:
Avaya strongly recommends that you leave the default values for the following screen.

Four parameters on the maintenance-related system-parameters screen determine threshold values for network performance. You can use the default values for these parameters, or you can have an Avaya representative change them to fit the needs of your network. These threshold values apply only to IP trunks, and do not effect other IP endpoints.

The craft (or higher) login is required to perform this administration. To administer the maintenance-related system-parameters screen do the following:

1. Type **change system-parameters ip-options** to open the ip-options system-parameters screen.

```
change system-parameters ip-options              Page 1 of 1
               IP-OPTIONS SYSTEM PARAMETERS


IP MEDIA PACKET PERFORMANCE THRESHOLDS
  Roundtrip Propagation Delay (ms)    High: 800   Low:400
                  Packet Loss (%)     High: 40        Low:15
                  Ping Test Interval (sec): 20
  Number of Pings Per Measurement Interval: 10
```

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

Field	Conditions/Comments
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. Submit the screen.

These parameters have no effect unless the **bypass if IP threshold exceeded** field is set to **y** on the Signaling Group screen. 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. They 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

Quality of Service to Endpoints

QoS parameter values are provided by the endpoint installer, the user, or DHCP. If VLAN information is later sent to the endpoint by the Avaya MultiVantage software, this information is ignored by the endpoint. However, if the system sends priority information, then the endpoints will use the updated information and ignore the original parameters set by the installer, user, or DHCP.

DiffServ and 802.1p priorities apply to both ends of a connection. Configure QoS values for the IP interfaces and IP endpoints to correspond to the network to which they are connected. In many cases, the endpoints are “local” and the values for the endpoints and the networks are the same. In other cases, the endpoints and the network have different QoS values. The system can download default values to the endpoint when values are not provided by the endpoint installer or the user.

Quality of Service Integration with APM

Avaya MultiVantage allows you more control over Quality of Service (QoS) policies in your IP voice network by using Avaya Policy Manager (APM) to establish Quality of Service parameters. Use of APM ensures that policies are consistently applied to both the data network and the voice network.

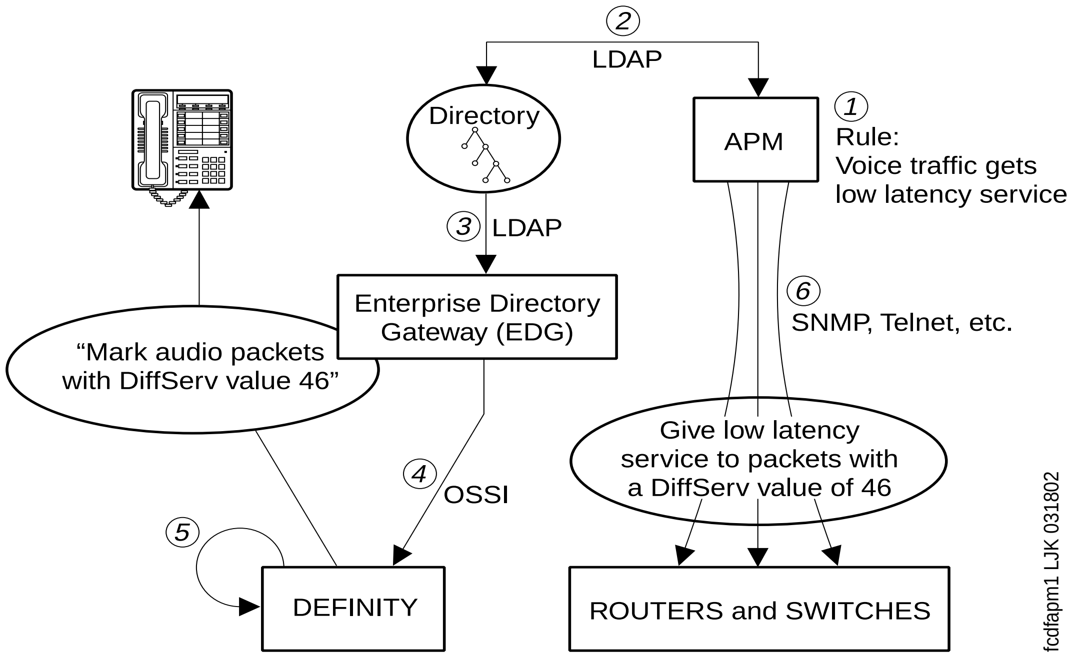
APM provides a central repository for QoS policies and allows comprehensive QoS management across routers, switches, and endpoints. QoS policies are assigned according to network regions and are distributed through the Enterprise Directory Gateway to your systems and to routers and switching devices.

APM is a policy management system that allows network managers to define network utilization policies. These policies are targeted on elements in an enterprise directory, such as users, groups, and applications, as well as other factors such as time of day, and can therefore express a rich variety of business rules. These policies can be uniformly and consistently applied throughout the network. APM is driven by a directory and can communicate its policy based rules to various data switches and routers (including Cajun and Cisco®).

Avaya Policy Manager Release 2.1 or later includes support for the management of QoS parameters introduced in DEFINITY Release 9. APM is able to communicate these QoS policy-based rules to the system via the Enterprise Directory Gateway. The Enterprise Directory Gateway contains LDAP schema for Avaya MultiVantage administration data and communicates with Avaya MultiVantage solutions via the proprietary OSSI protocol. The LDAP schema in the Enterprise Directory Gateway is extended to include the QoS parameters described above.

APM is able to populate the Avaya MultiVantage-specific QoS LDAP schema in the Enterprise Directory Gateway. The Enterprise Directory Gateway then sends the updated QoS parameters to the Avaya MultiVantage solution via the OSSI protocol. APM also communicates these same QoS policy rules to the various data switches and routers in the network, so that there can be consistent quality of service across the network for the voice packets.

The following figure illustrates this architecture:



Scenario

1. Business rule established in Avaya Policy Manager (APM)

2. APM uses LDAP to update Avaya MultiVantage solution schema in the directory.

3. Enterprise Directory Gateway (EDG) identifies the change in the directory.

4. EDG updates the Avaya MultiVantage solution administration via the OSSI protocol.

5. The Avaya MultiVantage solution tells the Media Processor, CLAN, and IP Phones to mark audio packets with DSCP=46.

6. APM distributes policy information to other network devices, including low latency service for DiffServ value of 46.

Avaya VoIP Monitoring Manager

The Avaya VoIP Monitoring Manager, a VoIP Network Quality monitoring tool, is available for customers. It is able to monitor many network quality affecting factors, including levels of jitter, packet loss, delay, codec used, and RSVP status. For more information about this tool, please see your Avaya representative.

QSIG

5

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. QSIG complies with the International Organization for Standardization (ISO) 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

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

Avaya MultiVantage provides different levels of QSIG functionality. You can view the status of each level on the System Parameters Customer Options screen.

The following table lists the QSIG features supported by Avaya MultiVantage 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 MultiVantage systems (see note below).

QSIG Category	Supported Features
Basic Call Setup	■ Basic Call Setup ■ Name and Number Transport ■ Transit Counter
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 MultiVantage 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).

NOTE:
Although VALU-MSI only works between Avaya MultiVantage systems, MSI information will be 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 will still arrive at each end. Similarly, if two non-Avaya systems are sending their own MSI through an Avaya MultiVantage switch, and the connections are all QSIG, the Avaya MultiVantage switch will send on the information.

Basic call setup

Transit counter (TC)

Avaya MultiVantage 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 Avaya MultiVantage 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 MultiVantage 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).

Example

If a non-restricted calling name and number are received by an Avaya MultiVantage 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.

Call Transfer

QSIG Call Transfer is based on the current Avaya MultiVantage 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 fuctionality 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). *Avaya MultiVantage uses only the first way.*

- 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 on 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, or
- 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 multiappearance 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:

1. Path reservation — there are two methods of establishing the CC call:
   a. path reservation method
   b. non-reservation method
2. Retention of signaling connection — there are two ways in which CC uses call independent signaling connections:
   a. connection retention method
   b. connection release method
3. Service retention — there are two possible behaviors when User B is found to be busy again after User A responds to CC recall:
   a. service retention method
   b. service cancellation method

As an originating switch for QSIG CC, Avaya MultiVantage will select 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, Avaya MultiVantage will select 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.

When an Intuity AUDIX system supports a QSIG network, the served user switch does not need to be a

As of Release 9, QSIG Centralized Intuity AUDIX has a new feature: Path optimization using QSIG Diversion with Reroute. Prior to Release 9, routing to a voice mail adjunct involved Diversion forward switching with no rerouting capability.

As of Release 11, Transfer into Intuity AUDIX will work 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.
- 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

- Unless you have 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 MultiVantage switch to be a served user switch of a non-Avaya MultiVantage message center switch. Therefore, when the messaging system is the message center switch, it can serve the Avaya MultiVantage served user 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 MultiVantage served user switch on all Avaya MultiVantage 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 via forward switching
- Gateway scenarios where Avaya MultiVantage, 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.

Avaya MultiVantage 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 the 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 adhered to 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 via 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 (page 77).

  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 (page 202), though the connectivity of the network itself differs. If administered, path replacement is invoked after coverage.

Centralized Attendant Services

Centralized Attendant Service (CAS)

The CAS feature enables one or more branches to concentrate their attendants on one main. CAS provides transparency between switches of most features that are normally available to the basic attendant service.

All current QSIG features are available with CAS. QISG-CAS does not interwork with RLT-CAS.

Potential CAS limitations

There are a few potential limitations when using CAS:

1. Path Replacement does not work immediately. This means that resources are being utilized longer with CAS.
2. Path Replacement is not guaranteed.
3. Path Replacement does not enable a branch to act as a gateway.
4. No path replacement functionality takes place during or after a conference.

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 the transfer.
- Attendant return call (release loop operation, returns to same attendant if available. If not, then 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 that Non-Call Associated Temporary Signaling Connections (NCA-TSCs) are.

NCA-TSC

An NCA-TSC is a connection not related with any ISDN B-channel connections. MultiVantage 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.

CAUTION:

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 the it active for a longer period of time.

Setting up QSIG

Follow these steps to set up QSIG:

1. Verify with your Avaya sales representative or project manager what QSIG capabilities the system should have. The following capabilities apply:

Table 6. QSIG capabilities

Capability categories	Cross-networking features
QSIG basic	■ Calling/connected name and number ■ Calling name and number identification ■ Transit Counter
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 MultiVantage solutions only)	■ Called/busy number display ■ Distinctive ringing ■ Call Coverage ■ Leave Word Calling
Centralized Attendant Service	■ Centralized Attendant

NOTE:
Although VALU-MSI only works between Avaya MultiVantage solutions, MSI information will be passed through non-Avaya MultiVantage solutions in an all QSIG network. Thus, if you have two Avaya MultiVantage switches connected using QSIG through a non-Avaya MultiVantage switch, the MSI information will still arrive at each end. Similarly, if two non-Avaya MultiVantage solutions are sending their own MSI through an Avaya MultiVantage switch, and the connections are all QSIG, the Avaya MultiVantage switch will send 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 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 2 and verify the following field:
   - **Async. Transfer Mode (ATM) Trunking** field is **y**.
6. On page 3, 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**.
7. On page 4, 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 MultiVantage solutions:

   - 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 MultiVantage solution 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 - 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

Setting up QSIG supplementary services

Follow these steps to set up QSIG supplementary services

1. Administer or check the ISDN Numbering - Public/Unknown screen.
2. Administer or check the ISDN Numbering - Private screen.
3. Administer or check the signaling group screen. 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. 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 - **y**
   - QSIG TSC Extension - valid extension number to serve as TSC for both incoming and outgoing QSIG network calls.
   - 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
   - (For AUDIX only) MWI - Number of Digits per AUDIX Subscriber - 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.

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 Call Completion, Call Offer, and Call Forward:
   - Restrict Call Forward Off-Net - n
   - Auto Callback- y
   - QSIG Call Offer Originations - y

Call Completion

In addition to the Basic QSIG Supplementary Services administration described above, complete the following administration:

- On the Trunk Group screen, page 1, set the Supplementary Service Protocol field to **b** and administer the trunk for Call Independent Signalling Connections.

Transfer into Avaya QSIG Message Center

This feature works with Octel Serenade only. In addition to the Basic QSIG Supplementary Services administration described above, complete the following administration:

- On the System-Parameters Customer-Options screen, page 6, the Transfer Into QSIG Voice Mail field must be set to **y**.
- On the Feature Access Code (FAC) screen, page 3, assign a Feature Access Code in the Transfer to Voice Mail Access Code field. (This field is called “Transfer to AUDIX Access Code” in pre-R8 systems).
- 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 R9 or later software. Also, the Interworking with DCS field on the QSIG Optional Features screen 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.

Setting Up Centralized Attendant Services

Follow these steps to set up 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 MultiVantage Software* for instructions on administering an attendant console.

1. Enable QSIG Supplementary Services with Rerouting on the System Parameters Customer Options screen, page 6, as described above.
2. On the System Parameters Customer Options screen, page 6, 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 for QSIG CAS Number.
    b. If **QSIG-branch** is entered in the CAS field, then the field AAR/ARS Access Code is optional.
4. Administer the QSIG ISDN trunk groups screen with option **b** for Supplementary Service Protocol.
5. Assign an extension to Attd on the Dial Plan Analysis screen at the main.
6. Administer each QSIG Supplementary Service that will be used by attendants.

Setting Up QSIG VALU Call Coverage

Follow these steps to set up QSIG VALU Call Coverage.

1. Enable the QSIG Basic Supplementary Services on the Customer Options screen, page 6, described above.
2. Enable (enter **y**) Value-Added (VALU) on the System-Parameters Customer-Options screen, page 6, as described above.
3. On a Trunk Group screen, enter **y** in the QSIG Value-Added field, page 1, and sent **b** in the supplementary Service Protocol Option field, page 1.

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 MultiVantage 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 (via Station screen) takes precedence over the system-wide “coverage after forwarding” activation/deactivation selection (via the System Parameters Call Coverage/Call Forwarding screen). With QSIG Diversion with Rerouting active, the system-wide selection takes precedence unless you enter **y.**

   See the examples in the following table:

Cvg. After Fwd (Station Screen)	Cvg. After Fwd (Sys Params. Coverage Screen)	QSIG VALU Coverage Overrides QSIG Diversion	Then
y	n	n	Call doesn’t 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.
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.

NOTE:
If Maintain SBA at Principal 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. See also the Remote Call Coverage Table screen in the Administrator’s Guide.
6. Define coverage paths for users as required.

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.

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

Related Administration of Terminating Extension Groups

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

Related Administration of AUDIX/Message Centers

Follow these steps to set up Related Administration of AUDIX/Message Centers.

NOTE:
Set up QSIG TSCs before you administer messaging. See "Call Completion".

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 CLAN 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**
   - 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 MultiVantage solution, the digits entered should reroute the call to the AUDIX hunt group on the message center switch.
   - Calling Party Number to AUDIX? - **y**

QSIG Interactions

Consider the following feature interactions when configuring QSIG on your system.

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. This is an Avaya MultiVantage solution only, MSI solution.

- 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 Cover after Forward 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 interactions

- Distributed Communications Systems (DCS+)

  In a DCS+ network, MultiVantage will display 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, MultiVantage 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.

  MultiVantage interworks called/calling/connected name and number identification between DCS+ and QSIG.

Path Replacement

- 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:
The Avaya MultiVantage solution 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

  RR 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

- 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

  TC does not apply to TAC calls.

Call Completion

- 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 *yes*.
- 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 Indications

- 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 TSCs used for QSIG MWI uses FRL 7.

- 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

  QSIG MWI messages use the default COR of unrestricted.
- Class of Service

  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 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 (page 202)

- 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 utilize Remote Call Coverage table and QSIG-VALU Call Coverage is supported only for coverage points associated with 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 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 (Temporary Bridge Appearance) is not maintained for the Principal user (i.e. 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 (i.e. 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 the trunk resources and provide path optimization without QSIG Path Replacement.

Centralized Attendant Service (CAS)

- 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 privately call another party 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 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 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 realerts 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 admin check to block such administration, but QSIG CAS may not function correctly.

- Automatic Circuit Assurance

  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 an 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 via 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

  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 an 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.

  MultiVantage solutions support 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 go on and handle other calls), Path Replacement is attempted.

- Security Violation Notification

  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 which attendant group the priority level routes to and whether calls of that priority level alert with emergency tone.

- Special Application 8141 - 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, via 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.

Migration to QSIG considerations

If you are planning to migrate your network from DCS to QSIG, then there are some issues which you will need to consider. The following is a list of some of the issues you might consider:

- Feature Parity
- Virtual Private Networking Facilities
- 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.

Systems with Avaya MultiVantage Software have MSI features that emulate DCS features not standard with QSIG.

The following table compares DCS and QSIG features supported on MultiVantage solutions:

Capabilities	DCS			QSIG	
	DCS	DCS with Path Replacement	DCS with call coverage	Basic	supplementary
Voice/Data Calls (Bearer type)	(DCS+)			●	
Called Name Display for terminals	*				●
Called Number	*				●
Calling Name Display	*				●
Calling Number Display	*			●	
Connected Name Display	* (DCS+)				●
Connected Number Display	(DCS+)			●	
Attendant Display	*				●
Automatic Call Back	*				●

Capabilities	DCS			QSIG	
Automatic Circuit Assurance	*				
Busy verification of Term/Trunks	*				
Call Coverage Across DCS Nodes			* (DCS+)		
C/F Diversion unconditional		*			●
C/F Diversion Busy/No Reply	*				●
C/F Diversion W/Reroute		*			●
Call Independent Signaling Connections, Temporary Signaling Connections	(DCS+)				●
Call Transfer	*				●
Call Transfer with Rerouting		(DCS+)			●
Call Waiting (Offer)	*				●
Centralized Intuity AUDIX	*				●
Priority & Distinctive Ringing	*				●
Leave Word Calling	*				●
Message Waiting Indication	*				●
Path Retention	*				●
Transit Counter				●	
QSIG Voice Mail	*				●
Call Coverage across QSIG networks	*				●
Transfer into Serenade QSIG Voice Mail					●
Path Optimization after Transfer Out of Voice Mail		(DCS+)			●

* Denotes traditional DCS

(DCS+) Denotes DCS over ISDN PRI

● Denotes QSIG Basic or Supplementary

Not shown in the above table is QSIG Centralized Attendant Service (CAS) with MSI. There is no DCS/DCS+ CAS offering.

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 with MultiVantage software. 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 have R11 or newer software. LWC will activate MWI lamps on MultiVantage stations only.

A Complex Networking Example

6

This chapter provides an example of a complex network. It describes procedures for administering trunk groups, dial plans, signaling groups, and data links for a four-switch network with an Intuity AUDIX and a CMS.

The network example is unchanged from the example in Issue 1 (for R7) of this book. The screens have been updated for R11.

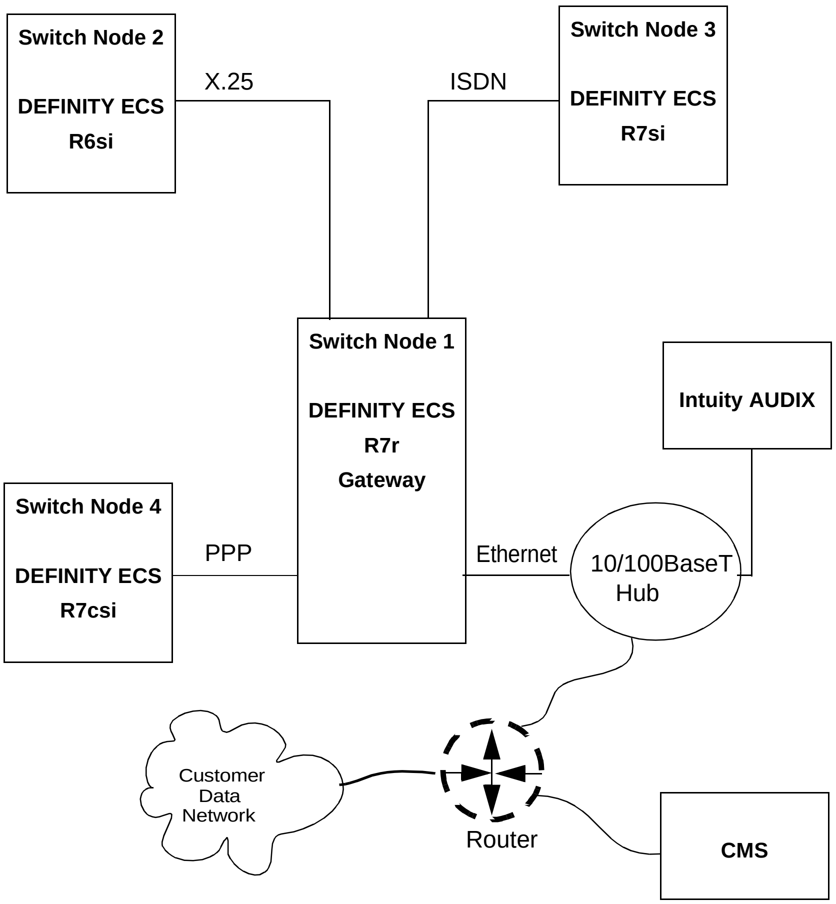
Overview

This section shows a high-level diagram of the example network and lists the administration tasks that need to be completed for each node to set up the network.

NOTE:
The term node defines a network interface such as a port on the C-LAN board. A "DCS node" is referred to as a "Switch Node." Thus, a Switch Node (a switch) can have many nodes (network interfaces). The Dial Plan and AAR Digit Analysis Table screens have fields that use node to refer to a switch.

Network Diagram

The following diagram shows a high-level view of the example network.



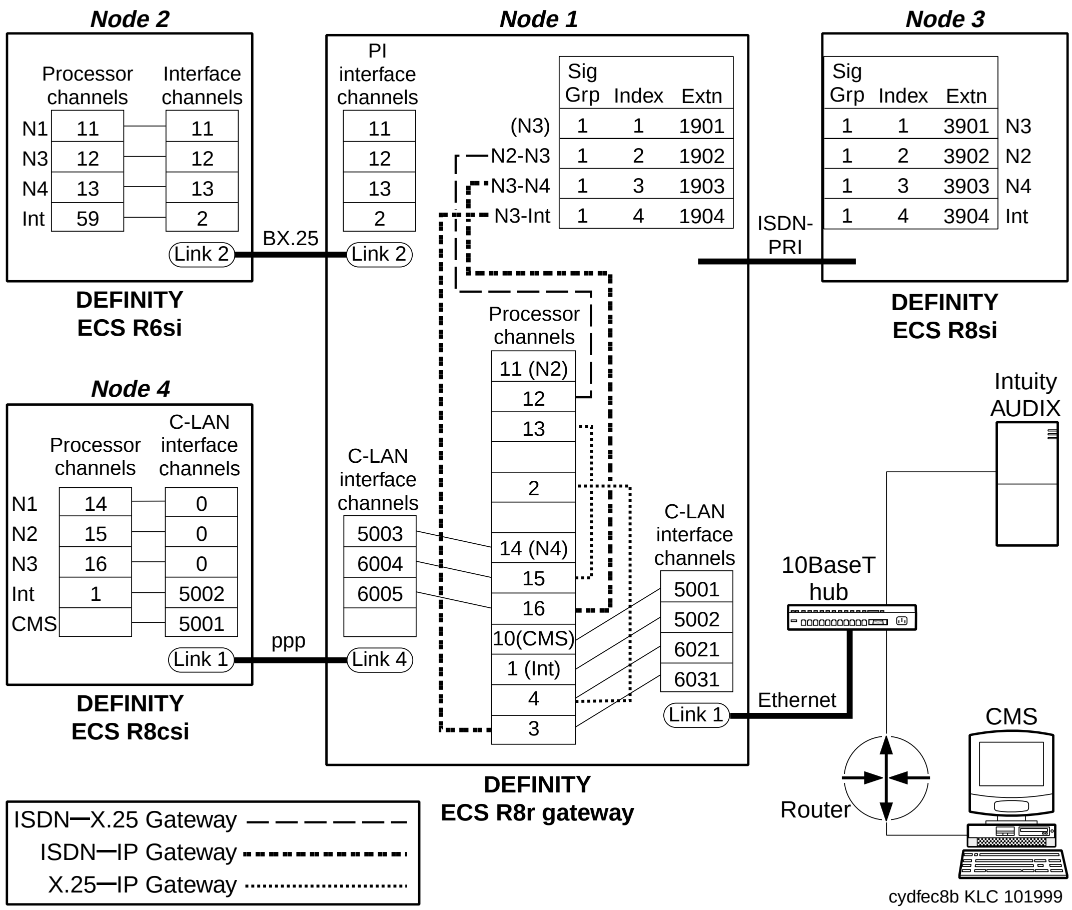
Task Summary

The following matrix summarizes the administration tasks required to set up this network.

Administration Task	Switch Node 1 R7r	Switch Node 2 R6si	Switch Node 3 R7csi	Switch Node 4 R7si	CMS	Intuity AUDIX
DS1 Circuit Pack	X	X	X	X		
Synchronization Plan	X	X	X	X		
Trunk Groups	X	X	X	X		
Dial Plan	X	X	X	X		
Uniform Dialing Plan	X	X	X	X		
AAR Digit Analysis	X	X	X	X		
Signaling Group	X		X			
Hunt Group		X	X	X		
ISDN TSC Gateway Channel Assignment	X					
Interface Links		X				
Node Names	X			X	X	X
IP Route	X			X	X	X
Data Modules	X	X	X	X		
Processor Channel Assignments	X	X	X	X	X	X
Routing Patterns	X	X	X	X		
Bus Bridge			X			
Intuity Translations for DCS Audix						X
CMS Server Translations					X	

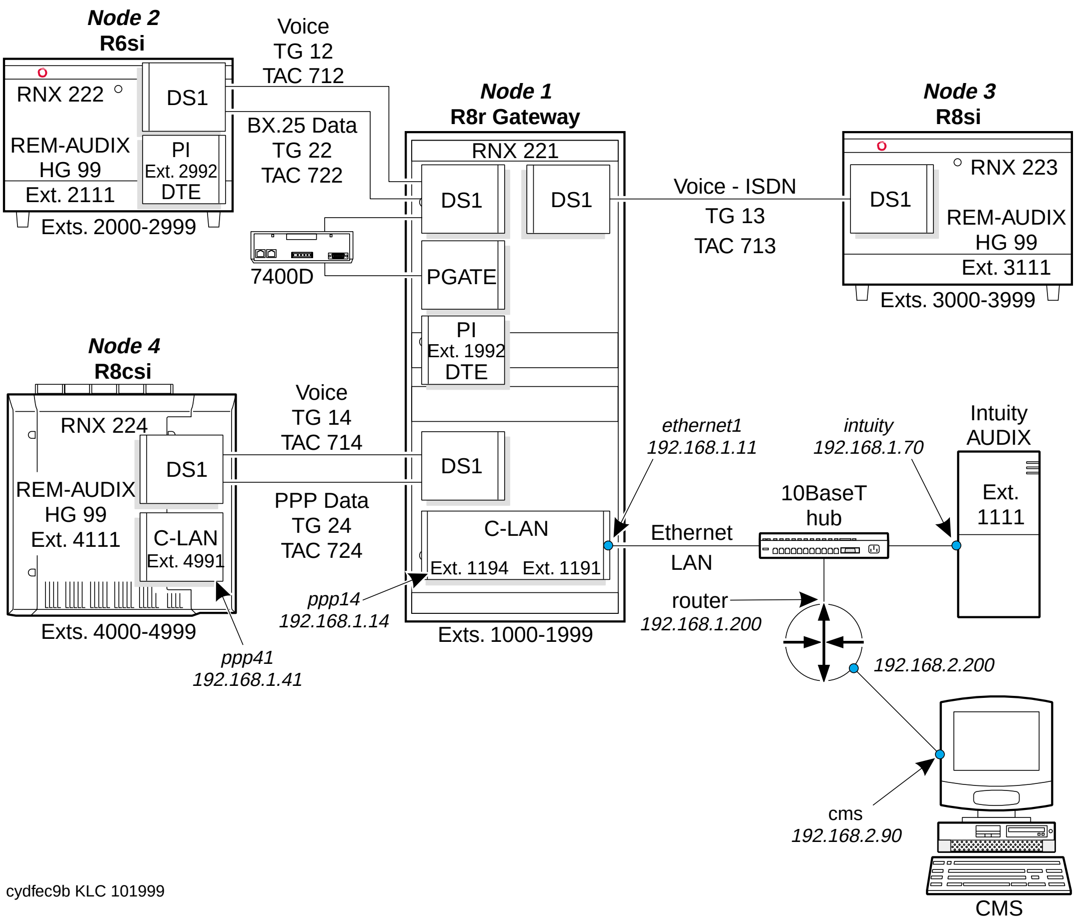
Link/Channel/TSC Map

The link/channel/TSC map shows the link, processor channel, and temporary signaling connection numbers, and the connection types for each Switch Node in the network.



Network Map

The network map shows the RNX, trunk group, hunt group numbers, data module type, and extension for each Switch Node in the network. For TCP/IP connections, the node name and IP address is shown for each node.



cydfec9b KLC 101999

Switch-Node 1 Administration

This section displays the screens for the administration of Switch Node 1.

DS1 Circuit Packs

Connection to Switch Node 2

```
add ds1 1c19                                                  Page   1 of   2
                                 DS1 CIRCUIT PACK

            Location: 01c19                           Name: Switch 1 to Switch 2
            Bit Rate: 1.544                    Line Coding: b8zs
   Line Compensation: 1                       Framing Mode: esf
      Signaling Mode: common-chan



Interface Companding: mulaw
           Idle Code: 11111111
             DMI-BOS? n




      Slip Detection? n                 Near-end CSU Type: other
```

Connection to Switch Node 3

```
add ds1 1c18                                                  Page   1 of   2
                                 DS1 CIRCUIT PACK

            Location: 01c18                           Name: Switch 1 to Switch 3
            Bit Rate: 1.544                    Line Coding: b8zs
      Signaling Mode: isdn-pri
             Connect: pbx                        Interface: network
   TN-C7 Long Timers?                     Country Protocol: 1
Interworking Message:                     Protocol Version: a
Interface Companding: mulaw                           CRC? n
           Idle Code: 11111111
                             DCP/Analog Bearer Capability: 3.1kHz



      Slip Detection? n                 Near-end CSU Type: other
```

Connection to Switch Node 4

```
add ds1 1c17                                                      Page   1 of   2
                                DS1 CIRCUIT PACK

            Location: 01C17                             Name: Switch 1 to Switch 4
            Bit Rate: 1.544                      Line Coding: b8zs
   Line Compensation: 1                          Framing Mode: esf
      Signaling Mode: robbed bit


Idle Code: 11111111



      Slip Detection? n
```

Dial Plan Analysis

```
change dialplan analysis                                   Page   1 of   3   SPE A
                             DIAL PLAN ANALYSIS TABLE
                                                              Percent Full:    9


      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
      String  Length Type      String  Length Type      String  Length Type
        0       1    attd
        1       4    dac
        2       4    ext
        3       4    ext
        4       4    ext
        7       3    dac
        8       1    fac
        9       1    fac
        *       3    dac
        #       3    dac
```

Dial Plan Parameters

```
change dialplan parameters                                              Page 1 of 1
                               DIAL PLAN PARAMETERS

      AAR/ARS Internal Call Prefix: ________             Local Node Number:  1
 AAR/ARS Internal Call Total Length: ___                   ETA Node Number:  __
                                                       ETA Routing Pattern:  __
             UDP Extension Search Order: local-extensions-first
          6-Digit Extension Display Format: xx.xx.xx
          7-Digit Extension Display Format: xxx.xxxx
```

Signaling Group

Page 1

```
add signaling-group next                                          Page 1 of 5
                               SIGNALING GROUP
Group Number: 1              Group Type:
                    Associated Signaling? y         Max number of NCA TSC: 5_
                       Primary D-Channel: 01c1824     Max number of CA TSC: 23
                                                  Trunk Group for NCA TSC: 13
       Trunk Group for Channel Selection: ___       Network Call Transfer?
          Supplementary Service Protocol: a__
```

Page 2 – Administered NCA TSC Assignment

```
add signaling-group next                                           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:   1901_     y     permanent   3901___________   dcs___  _______     3_
  2:   1902_     y     permanent   3902___________   gateway _______     __
  3:   1903_     y     permanent   3903___________   gateway _______     __
  4:   1904_     y     permanent   3904___________   gateway _______     __
  5:
  :
 16:
```

Synchronization Plan

```
change synchronization                                             Page 1 of X
                                 SYNCHRONIZATION PLAN
                    SYNCHRONIZATION SOURCE (circuit pack location)
                          Stratum: 4
_________________________Primary: 01c20      Secondary: ___

       Location Type   Name Slip  Location  Type   Name             Slip
       01c17 DCS Node 4_____  n    UDS1-BD  _____  _______________   _
       01c18 DCS Node 3____   n    TBRI-BD  _____  _______________   _
       01c19 DCS Node 2_____  n             _____  _______________   _
       01c20 AT&T ___________ y             _____  _______________   _
       _____ _______________  _             _____  _______________   _
       _____ _______________  _             _____  _______________   _
       _____ _______________  _             _____  _______________   _
       _____ _______________  _             _____  _______________   _     ___
NOTE: TN722B & TN464B DS1 sources result in stratum 4, type II synchronization
```

Trunk Groups

Group 12 (tie to Switch Node 2) — page 1

```
add trunk-group 12                                              Page   1 of  10
                                TRUNK GROUP

Group Number: 12                   Group Type: tie          CDR Reports: y
  Group Name: Switch 1 to Switch 2 tg12   COR: 1        TN: 1        TAC: 712
   Direction: two-way        Outgoing Display? n    Trunk Signaling Type:
 Dial Access? y              Busy Threshold: 99         Night Service:
Queue Length: 0                                     Incoming Destination:
   Comm Type: avd                    Auth Code? n
                                   Trunk Flash? n
         BCC: 0
TRUNK PARAMETERS
   Trunk Type (in/out): wink/wink        Incoming Rotary Timeout(sec): 5
    Outgoing Dial Type: tone                        Incoming Dial Type: tone
      Wink Timer(msec): 300                  Disconnect Timing(msec): 500
       Digit Treatment:                                         Digits: __
                                                    Sig Bit Inversion: none
     Connected to Toll? n      STT Loss: normal       DTT to DCO Loss: normal
    Incoming Dial Tone? y
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0            Receive Answer Supervision? y
```

Group 12 — Page 2

```
add trunk-group 12                                              Page   2 of  10
TRUNK FEATURES
         ACA Assignment? n           Measured: none
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n
                                 Glare Handling: none
           Used for DCS? y   PBX ID: 2
   Suppress # Outpulsing? n
                     Seize When Maintenance Busy: neither-end



Incoming Tone (DTMF) ANI: no                   Per Call CPN Blocking Code:
         Connected to CO? n                  Per Call CPN Unblocking Code:
```

Group 12 — member assignments

```
add trunk-group 12                                             Page 3 of x
                              TRUNK GROUP
                                      Administered Members (min/max):   1/3
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   3

       Port    Code Sfx Name        Night            Mode      Type   Ans Delay
  1: 01c1901  TN767  F                                                   20
  2: 01c1902  TN767  F                                                   20
  3: 01c1903  TN767  F                                                   20
  4:
  5:
```

Group 22 (data to Switch Node 2)— page1

```
add trunk-group 22                                          Page   1 of  11
                                TRUNK GROUP

Group Number: 22                   Group Type: tie          CDR Reports: y
  Group Name: DCS data to node 2 - TG22   COR: 1        TN: 1        TAC: 722
   Direction: two-way       Outgoing Display? n    Trunk Signaling Type:
 Dial Access? y               Busy Threshold: 99          Night Service:
Queue Length: 0                                    Incoming Destination:
   Comm Type: data                 Auth Code? n
                                 Trunk Flash? n

TRUNK PARAMETERS
   Trunk Type (in/out): wink/wink       Incoming Rotary Timeout(sec): 5
    Outgoing Dial Type: tone                      Incoming Dial Type: tone
                                             Disconnect Timing(msec): 500
       Digit Treatment:                                       Digits: __
                                                   Sig Bit Inversion: none
     Analog Loss Group:_______                    Digital Loss Group:
    Incoming Dial Tone? y

 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0            Receive Answer Supervision? y
```

Group 22 — Page 2

```
add trunk-group 22                                                 Page   2 of  11
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n

            Used for DCS? n
   Suppress # Outpulsing? n
                    Seize When Maintenance Busy: neither-end




Incoming Tone (DTMF) ANI: no                      Per Call CPN Blocking Code:
         Connected to CO? n                     Per Call CPN Unblocking Code:
                                                       Ds1 Echo Cancellation?
```

Group 22 — Group member assignments

```
add trunk-group 22                                                     Page 5of x
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/3
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   3

       Port    Code Sfx Name        Night           Mode      Type   Ans Delay
  1: 01c1923  TN767  F                                                   20
  2:
  3:
  4:
  5:
  6:
  7:
  8:
  9:
 10:
```

Group 13 (ISDN-PRI to Switch Node 3) — page 1

```
add trunk-group 13                                              Page   1 of  10
                                TRUNK GROUP

Group Number: 13                   Group Type: isdn         CDR Reports: y
  Group Name: ISDN TG 13 to Switch 3      COR: 1        TN: 1        TAC: 713
   Direction: two-way       Outgoing Display? n         Carrier Medium: _
 Dial Access? n               Busy Threshold: 99         Night Service: _____
Queue Length: 0
Service Type: tie                   Auth Code? n          TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send TCM,Lookahead: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: a     Digit Handling (in/out): enbloc/enbloc

             Trunk Hunt: cyclical             QSIG Value Added?
      Connected to Toll? n                  Digital Loss Group:
Calling Number - Delete:          Insert:             Numbering Format:
               Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
```

Group 13 — page 2

```
add trunk-group 13                                              Page   2 of  10
                                TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                               Data Restriction? n       Maintenance Tests? y
                                      Send Name: n     NCA-TSC Trunk Member: 1
                                                        Send Calling Number: n
            Used for DCS? y
              Suppress # Outpulsing? n
 Outgoing Channel ID Encoding: preferred        UUI IE Treatment: service-provider
                                      Replace Restricted Numbers?
                                      Replace Unavailabe Numbers?
                                           Send Connected Number: n

             Send UUI IE?                                    Ds1 Echo
Cancellation?
              Send UCID? n                        US NI Delayed Calling Name Update?
 Send Codeset 6/7 LAI IE? y       Network (Japan) Needs Connect Before Disconnect?
```

Group 13 — member assignments

```
add trunk-group 13                                              Page   4 of  10
                                 TRUNK GROUP
                                      Administered Members (min/max):   0/0
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   0

       Port    Code Sfx Name        Night            Sig Grp
  1: 01c1801  TN464  E                                 1
  2: 01c1802  TN464  E                                 1
  3: 01c1803  TN464  E                                 1
  4: 01c1804  TN464  E                                 1
  5:
  6:
  7
```

Group 14 (tie to Switch Node 4) — page 1

```
add trunk-group 14                                               Page   1 of  11
                                TRUNK GROUP

Group Number: 14                   Group Type: tie           CDR Reports: y
  Group Name: Switch 1 to Switch 4 tg14   COR: 1        TN: 1        TAC: 714
   Direction: two-way      Outgoing Display? n    Trunk Signaling Type:
 Dial Access? y              Busy Threshold: 99          Night Service:
Queue Length: 0                                   Incoming Destination:
   Comm Type: voice                Auth Code? n
                                 Trunk Flash? n
         BCC: 0
TRUNK PARAMETERS
   Trunk Type (in/out): wink/wink        Incoming Rotary Timeout(sec): 5
    Outgoing Dial Type: tone                       Incoming Dial Type: tone
       Digit Treatment:                       Disconnect Timing(msec): 500
                                                               Digits: __
                                                    Sig Bit Inversion: none
     Analog Loss Group:                            Digital Loss Group:
    Incoming Dial Tone? y
                                      Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0            Receive Answer Supervision? y
```

Group 14 — Page 2

```
add trunk-group 14                                                 Page   2 of  11
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n

            Used for DCS? y
   Suppress # Outpulsing? n
                    Seize When Maintenance Busy: neither-end




Incoming Tone (DTMF) ANI: no                     Per Call CPN Blocking Code:
         Connected to CO? n                    Per Call CPN Unblocking Code:
                                                      Ds1 Echo Cancellation?
```

Group 14 — member assignments

```
 TRUNK GROUP                           Page 5 of 11
                                        Administered Members (min/max):   1/3
GROUP MEMBER ASSIGNMENTS                    Total Administered Members:   3

       Port    Code Sfx Name         Night           Mode      Type   Ans Delay
  1: 01c1701 TN767  F                                                   20
  2: 01c1702 TN767  F                                                   20
  4: 01c1703 TN767  F                                                   20
  5:
  6:
  7:
  8:
  9:
 10:
```

Group 24 (data to Switch Node 4) — page1

```
add trunk-group 24                                             Page   1 of  10
                                TRUNK GROUP

Group Number: 24                  Group Type: tie          CDR Reports: y
  Group Name: DCS data to node 2 - TG22  COR: 1        TN: 1        TAC: 722
   Direction: two-way       Outgoing Display? n    Trunk Signaling Type:
 Dial Access? y               Busy Threshold: 99          Night Service:
Queue Length: 0                                    Incoming Destination:
   Comm Type: data                Auth Code? n
                                Trunk Flash? n

TRUNK PARAMETERS
   Trunk Type (in/out): wink/wink        Incoming Rotary Timeout(sec): 5
    Outgoing Dial Type: tone                        Incoming Dial Type: tone
                                               Disconnect Timing(msec): 500
       Digit Treatment:                                         Digits: __
                                                     Sig Bit Inversion: none
     Analog Loss Group:_______                      Digital Loss Group:
    Incoming Dial Tone? y

 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0          Receive Answer Supervision? y
```

Group 24 — Page 2

```
add trunk-group 24                                             Page   2 of  10
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n

            Used for DCS? n
   Suppress # Outpulsing? n
                    Seize When Maintenance Busy: neither-end



Incoming Tone (DTMF) ANI: no                Per Call CPN Blocking Code:
         Connected to CO? n               Per Call CPN Unblocking Code:
                                                 Ds1 Echo Cancellation?
```

Group 24 — member assignments

```
TRUNK GROUP                          Page 5 of 11
                                         Administered Members (min/max):   1/3
GROUP MEMBER ASSIGNMENTS                     Total Administered Members:   3

       Port    Code Sfx Name         Night           Mode      Type   Ans Delay
  1: 01c1723  TN767  F                                                   20
  2:
  3:
  4:
  5:
  6:
  7:
  8:
  9:
 10:
```

Uniform Dial Plan

NOTE:
Uniform Dial Plan administration changes considerably with R11. Now you use the Uniform Dial Plan Table to administer UDP. See the *Administrator's Guide for Avaya MultiVantage Software* for more information.

```
change uniform-dialplan 0                                               Page 1 of 2
                         UNIFORM DIAL PLAN TABLE
                                                   Percent Full:   2

Matching             Insert          Node     Matching          Insert          Node
Pattern   Len Del    Digits Net Conv Num      Pattern   Len Del Digits Net Conv Num
2          4   0     817    aar  n            ______    __  __  ____   __  __   __
4          5   1     334    aar  n            ______    __  __  ____   __  __   __
43659      5   1     928    aar  y            ______    __  __  ____   __  __   __
623        3   3     5380   aar  n            ______    __  __  ____   __  __   __
73012      5   1            enp  n     31     ______    __  __  ____   __  __   __
74100      5   0     81     ars  y            ______    __  __  ____   __  __   __
8          5   0            ext  n            ______    __  __  ____   __  __   __
911        3   0            ars  n            ______    __  __  ____   __  __   __
_____      _   _     _____  ___  _     __     ______    __  __  ____   __  __   __
_____      _   _     _____  ___  _     __     ______    __  __  ____   __  __   __
_____      _   _     _____  ___  _     __     ______    __  __  ____   __  __   __
_____      _   _     _____  ___  _     __     ______    __  __  ____   __  __   __
_____      _   _     _____  ___  _     __     ______    __  __  ____   __  __   __
```

AAR Digit Analysis

```
change aar analysis 1                                        Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                                          Percent Full:    6

          Dialed          Total     Route    Call   Node   ANI
          String         Min  Max  Pattern   Type   Num    Reqd
           222           7    7     102      aar    2      n
           223           7    7     103      aar    3      n
           224           7    7     104      aar    4      n
```

ISDN TSC Gateway Channel Assignment

```
change isdn tsc-gateway                                       Page   1 of   2
                      ISDN TSC GATEWAY CHANNEL ASSIGNMENT

    Sig    Adm'd NCA  Processor  Appli-     Sig   Adm'd NCA  Processor  Appli-
   Group   TSC Index   Channel   cation    Group  TSC Index   Channel   cation
  1: 1_       2_         12_     dcs__    17: __     __         ___     ______
  2: 1_       3_         16_     dcs__    18: __     __         ___     ______
  3: 1_       4_         3__     audix    19: __     __         ___     ______
  4: __       __         ___     _____    20: __     __         ___     ______
  :
```

Routing Patterns

Pattern 102

```
change route-pattern 102
                              Pattern Number: 102
                                No.                           DCS\
  Grp.  FRL NPA Pfx Hop Toll Del     Inserted              QSIG                 IXC
   No.            Mrk Lmt List Digits  Digits                Intw
 1:12
    0                      3                                              user
 2:                                                                          user
 3:                                                                          user
 4:                                                                          user
 5:                                                                          user
 6:                                                                          user
                                                                     No.
    BCC VALUE  TSC CA-TSC    ITC  BCIE Service/Feature  BAND   Dgts      Numbering
LAR
   0 1 2 3 4 W    Request                                  Subaddress        Format
 1: y y y y y n  n            both ept
                                          none
 2: y y y y y n  n           rest                                                none
 3: y y y y y n  n           rest                                                none
 4: y y y y y n  n           rest                                                none
 5: y y y y y n  n           rest                                                none
 6: y y y y y n  n           rest                                                none
```

Pattern 103

```
change route-pattern 103
                            Pattern Number: 103
                              No.                            DCS\
  Grp.  FRL NPA Pfx Hop Toll Del     Inserted              QSIG                IXC
   No.             Mrk Lmt List Digits  Digits                Intw
 1:12
    0                     3                                                 user
 2:                                                                            user
 3:                                                                            user
 4:                                                                            user
 5:                                                                            user
 6:                                                                            user
                                                                 No.
    BCC VALUE  TSC CA-TSC    ITC  BCIE Service/Feature  BAND   Dgts      Numbering
LAR
   0 1 2 3 4 W     Request                               Subaddress        Format
 1: y y y y y n  n             both ept
                                        none
 2: y y y y y n  n           rest                                              none
 3: y y y y y n  n           rest                                              none
 4: y y y y y n  n           rest                                              none
 5: y y y y y n  n           rest                                              none
 6: y y y y y n  n           rest                                              none
```

Pattern 104

```
change route-pattern 104
                            Pattern Number: 104
                              No.                            DCS\
  Grp.  FRL NPA Pfx Hop Toll Del     Inserted              QSIG                IXC
   No.             Mrk Lmt List Digits  Digits                Intw
 1:12
    0                     3                                                 user
 2:                                                                            user
 3:                                                                            user
 4:                                                                            user
 5:                                                                            user
 6:                                                                            user
                                                                 No.
    BCC VALUE  TSC CA-TSC    ITC  BCIE Service/Feature  BAND   Dgts      Numbering
LAR
   0 1 2 3 4 W     Request                               Subaddress        Format
 1: y y y y y n  n             both ept
                                        none
 2: y y y y y n  n           rest                                              none
 3: y y y y y n  n           rest                                              none
 4: y y y y y n  n           rest                                              none
 5: y y y y y n  n           rest                                              none
 6: y y y y y n  n           rest                                              none
```

Node Names

Audix

```
change node-names audix-msa                                         Page   1 of   1
                              AUDIX-MSA NODE NAMES

      Audix Names   IP Address          MSA Names    IP Address
      intuity_      192.168.200.10_     msa____      ___.___.___.___
```

IP

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

   Name               IP Address          Name                 IP Address
ppp14___________   192.168.200.12_    _______________    ___.___.___.___
ppp41___________   192.168.200.14_    _______________    ___.___.___.___
CMS_____________   192.168.201.10_    _______________    ___.___.___.___
router__________   192.168.200.1__    _______________    ___.___.___.___
Ethernet1_______   192.168.200.11_
.
```

Data Modules

x.25 data module

```
add data-module 1992
                                                                Page 1 of 2
                              DATA MODULE

Data Extension: 1992                  Name: x.25 on link 2 to node 2
          Type: x.25                        Remote Loop-Around Test? n
          Port: 01C0101       COR: 1             Destination Number: external
     Baud Rate: 9600           TN: 1           Establish Connection? n
 Endpoint Type: adjunct                      Connected Data Module: 1900
          Link: 2         DTE/DCE: dte                Error Logging? n
                      Enable Link: n


 Permanent Virtual Circuit? y          Highest PVC Logical Channel: 64
  Switched Virtual Circuit? n
```

pdm data module

```
add data-module 1900                                               Page 1 of x
                          DATA MODULE

    Data Extension: 1900          Name:                              BCC: 2
              Type: pdm           COS: 1         Remote Loop-Around Test? n
              Port: 01c2001       COR:1            Secondary Data Module? n
    ITC restricted:                TN:1                     Connected To: dte

ABBBREVIATED DIALING
   List 1

SPECIAL DIALING OPTION

ASSIGNED MEMBER (Station with a data extension button for this data module)
         Ext             Name
     1.
```

PPP data module

```
add data-module 1994                                              Page 1 of x
                          DATA MODULE

    Data Extension: 1994            Name: _ppp on link 4 to node 4___  BCC: 2
              Type: PPP              COS: 1
              Port: 01c1502          COR: 1
              Link: 4_                TN: 1
       Enable Link? n

         Node Name: ppp14_____
       Subnet Mask: 255.255.255.0


  Establish Connection: y

DESTINATION
            Digits: 7241991_________
         Node Name: ppp41___________
             CHAP? n
```

Ethernet data module

```
add data-module 1191                                                  Page 1 of X
                         DATA MODULE

   Data Extension: 1191          Name: Ethernet on link 1______________   BCC: 2
             Type: Ethernet
             Port: 01c1517_
             Link: 1_

 Network uses 1's for Broadcast Addresses? y
```

IP Interface

```
change ip-interfaces                                     Page   1 of   2

                                IP Interfaces

 Enable                                                                     Net
 Eth Pt Type   Slot  Code Sfx Node Name  Subnet Mask     Gateway Addr    Rgn
    y   C-LAN  01c15 TN799 B  Ethernet1  255.255.255.0__ 192.168.200.1__  1
    _   _____  _____ _____ _  _________  255.255.255.0   ___.___.___.___
    _   _____  _____ _____ _  _________  255.255.255.0   ___.___.___.___  _
    _   _____  _____ _____ _  _________  255.255.255.0   ___.___.___.___  _
    _   _____  _____ _____ _  _________  255.255.255.0   ___.___.___.___  _
    _   _____  _____ _____ _  _________  255.255.255.0   ___.___.___.___  _
    _   _____  _____ _____ _  _________  255.255.255.0   ___.___.___.___  _
    _   _____  _____ _____ _  _________  255.255.255.0   ___.___.___.___  _
```

Processor Channel Assignments

```
change communications-interface processor-channels
                                                                   Page 1 of X
                    PROCESSOR CHANNEL ASSIGNMENT

Proc                    Gtwy        Interface     Destination      Session    Mach
Chan Enable  Appl.      To  Mode    Link/Chan     Node        Port  Local/Remote  ID
  1:  y     audix           s       1    5002  intuity        0      1     1      1
  2:  y     gtwy-tcp 4      _       2    2     _________      _____  2     59     __
  3:  y     gateway         s       1    5031  intuity        0      3     3      __
  4:  y     gtwy-tcp 2      s       1    5021  intuity        0      2     2      __
  :
  :
 10:  y     mis             s       1    5001  cms            0      1     1      __
 11:  y     dcs             _       2    11    _________      _____  11    11     2
 12:  y     gateway         _       2    12    _________      _____  12    12     __
 13:  y     gtwy-tcp 15     _       2    13    _________      _____  13    13     __
 14:  y     dcs             s       4    5003  ppp41          0      14    14     4
 15:  y     gtwy-tcp 13     s       4    5004  ppp41          0      15    15     __
 16:  y     gateway         s       4    5005  ppp41          0      16    16     __
   :
 64:  _     _______         _       ___  ____  _________      0      ___   ___    __
```

Switch-Node 2 Administration

This section displays the screens for the administration of Switch Node 2.

DS1 Circuit Packs

```
add ds1 1a10
                                DS1 CIRCUIT PACK

            Location: 01a10                          Name: sw 2 to sw 1
            Bit Rate: 1.544                   Line Coding: b8zs
   Line Compensation: 1                      Framing Mode: esf
      Signaling Mode: common-chan


           Idle Code: 11111111
             DMI-BOS? n


      Slip Detection? n
```

Dial Plan Analysis

```
change dialplan analysis                                 Page   1 of   3   SPE A
                             DIAL PLAN ANALYSIS TABLE
                                                            Percent Full:    9


      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
      String  Length Type      String  Length Type      String  Length Type
        0       1    attd
        1       4    dac
        2       4    ext
        3       4    ext
        4       4    ext
        7       3    dac
        8       1    fac
        9       1    fac
        *       3    dac
        #       3    dac
```

Dial Plan Parameters

```
change dialplan parameters                                             Page 1 of 1
                              DIAL PLAN PARAMETERS

      AAR/ARS Internal Call Prefix: ________             Local Node Number:  2_
 AAR/ARS Internal Call Total Length: ___                   ETA Node Number:  __
                                                       ETA Routing Pattern:  __
             UDP Extension Search Order: local-extensions-first
         6-Digit Extension Display Format: xx.xx.xx
         7-Digit Extension Display Format: xxx.xxxx
```

Synchronization Plan

```
change synchronization                                          Page 1 of X
                                SYNCHRONIZATION PLAN
                   SYNCHRONIZATION SOURCE (circuit pack location)
                         Stratum: 4
                    Port Network: 1
                         Primary: 01a10      Secondary: ___

       Location   Name      Slip   Type     Location   Name          Slip  Type
       01a10 ______________  y    UDS1-BD   _____  _______________   _     ___
       _____ ______________  _              _____  _______________   _     ___
       _____ _______________ _              _____  _______________   _     ___
       _____ _______________ _              _____  _______________   _     ___
       _____ _______________ _              _____  _______________   _     ___
       _____ _______________ _              _____  _______________   _     ___
       _____ _______________ _              _____  _______________   _     ___
       _____ _______________ _              _____  _______________   _     ___
NOTE: TN722B & TN464B DS1 sources result in stratum 4, type II synchronization
```

Trunk Groups

Group 12 — page 1

```
add trunk-group 12                                            Page   1 of  10
                                TRUNK GROUP

Group Number: 12                   Group Type: tie          CDR Reports: y
  Group Name: DCS data to node 2 - TG22   COR: 1        TN: 1        TAC: 722
   Direction: two-way        Outgoing Display? n    Trunk Signaling Type:
 Dial Access? y                Busy Threshold: 99          Night Service:
Queue Length: 0                                     Incoming Destination:
   Comm Type: data                  Auth Code? n
                                  Trunk Flash? n

TRUNK PARAMETERS
   Trunk Type (in/out): wink/wink       Incoming Rotary Timeout(sec): 5
    Outgoing Dial Type: tone                      Incoming Dial Type: tone
                                             Disconnect Timing(msec): 500
       Digit Treatment:                                       Digits: __
                                                  Sig Bit Inversion: none
     Analog Loss Group:_______                  Digital Loss Group:
    Incoming Dial Tone? y

 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0          Receive Answer Supervision? y
```

Group 12 — Page 2

```
add trunk-group 12                                                 Page   2 of  10
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                 Internal Alert? n        Maintenance Tests? y
                               Data Restriction? n
                                 Glare Handling: none
            Used for DCS? y  PBX ID: 1
   Suppress # Outpulsing? n
                     Seize When Maintenance Busy: neither-end




Incoming Tone (DTMF) ANI: no                     Per Call CPN Blocking Code:
         Connected to CO? n                    Per Call CPN Unblocking Code:
```

Group 12 — member assignments

```
add trunk-group 12                                                 Page 4 of  10
                              TRUNK GROUP
                                      Administered Members (min/max):   1/3
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   3

       Port    Code Sfx Name        Night           Mode      Type   Ans Delay
  1: 01a1001  TN767  E
  2: 01a1002  TN767  E
  3: 01a1003  TN767  E
  4:
  5:
  6:
```

Uniform Dial Plan

NOTE:
Uniform Dial Plan administration changes considerably with R11. Now you use the Uniform Dial Plan Table to administer UDP. See the *Administrator's Guide for Avaya MultiVantage Software* for more information.

```
change uniform-dialplan 0                                                Page 1 of 2
                              UNIFORM DIAL PLAN TABLE
                                                                Percent Full:   2

Matching              Insert          Node      Matching          Insert          Node
Pattern   Len Del   Digits Net Conv Num       Pattern  Len Del Digits Net Conv Num
2_____     4   0    817_   aar  n    __       _____    __  __   ___   __   __  __
4_____     5   1    334_   aar  n    __       _____    __  __   ___   __   __  __
43659      5   1    928_   aar  y    __       _____    __  __   ___   __   __  __
623___     3   3    5380   aar  n    __       _____    __  __   ___   __   __  __
73012      5   1    ____   enp  n    31       _____    __  __   ___   __   __  __
74100      5   0    81__   ars  y    __       _____    __  __   ___   __   __  __
8_____     5   0    ____   ext  n    __       _____    __  __   ___   __   __  __
911___     3   0    ____   ars  n    __       _____    __  __   ___   __   __  __
_____      _   _    ____   ___  _    __       _____    __  __   ___   __   __  __
_____      _   _    ____   ___  _    __       _____    __  __   ___   __   __  __
_____      _   _    ____   ___  _    __       _____    __  __   ___   __   __  __
_____      _   _    ____   ___  _    __       _____    __  __   ___   __   __  __
_____      _   _    ____   ___  _    __       _____    __  __   ___   __   __  __
```

AAR Digit Analysis

```
change aar analysis 1                                          Page   1 of   2
                                  AAR DIGIT ANALYSIS TABLE
                                                              Percent Full:    6

          Dialed           Total     Route     Call    Node   ANI
          String         Min  Max   Pattern    Type    Num    Reqd
    221                  7    7      101       aar     1      n
    223                  7    7      101       aar     3      n
    224                  7    7      101       aar     4      n
```

Routing Patterns

```
change route-pattern 101
                              Pattern Number: 101

    Grp.  FRL NPA Pfx Hop Toll No. Del Inserted                             IXC
    No.           Mrk Lmt List Digits  Digits
  1:12     0                     3                                          user
  2:                                                                        user
  3:                                                                        user
  4:                                                                        user
  5:                                                                        user
  6:                                                                        user
```

Data Modules

procr-intf data module

```
add data-module 2992                                                Page 1 of 1
                                   DATA MODULE

  Data Extension: 2992_                    Name: x.25 on link 2 to node 1__
            Type: procr-intf          COS: 1       Maintenance Extension: 2002
Physical Channel: 02_                 COR: 1          Destination Number: 7221993
             ITC: Restricted           TN: 1       Establish Connection? y
          Link: 2_              DTE/DCE: dce      Connected Data Module: __________
                               Enable Link: n
ABBREVIATED DIALING
  List1: _______________

SPECIAL DIALING OPTION: ________________

ASSIGNED MEMBER (Station with a data extension button for this data module)

         Ext               Name
    1.
```

Processor Channel Assignments

Release 6 Processor Channel Assignment screen

```
change communications-interface processor-channels
                                                                      Page 1 of X
                  PROCESSOR CHANNEL ASSIGNMENT

    Proc             Interface                 Remote
    Chan    Appl.   Link  Chan     Priority     Proc  Chan         Machine-ID
     1:     _______  __   __    _________      ___   __              __
     :
    11:     dcs____   2   11    _________      _11   1_              __
    12:     dcs____   2   12    _________      _12   3_              __
    13:     dcs____   2   13    _________      _13   4_              __
      :
    59:     audix__   2   2_    _________      ___   __              __
```

Release 6 Interface Links screen

NOTE:
You must disable the link before you can make changes to the Interface Links screen.

```
change communications-interface links                            Page 1 of 1
                                INTERFACE LINKS
                 Est   PI                 Destination     DTE/
   Link  Enable  Conn   Ext      Port     Digits      Brd    DCE   Identification
    1:     _       _    _____    ____     __________  _____  ___   _______________
    2:     y       y    2992_    1a1004   1992 _____  _____  dce   _______________
    3:     _       _    _____    ____     __________  _____  ___   _______________
Link 1 [eia] - Connected to : ___ Clocking : ________
```

Hunt Group

page 1

```
add hunt-group 99                                                  Page   1 of  10
                                   HUNT GROUP

           Group Number: 99                                ACD? n
             Group Name: Intuity 1                       Queue? n
        Group Extension: 2111                           Vector? n
             Group Type: ucd-mia                  Coverage Path:
                     TN: 1         Night Service Destination:
                    COR: 1                  MM Early Answer? n
          Security Code:
    ISDN Caller Display:
```

page 2

```
add hunt-group 1                                                  Page   2 of  10
                                   HUNT GROUP

                           Message Center: rem-audix
                          AUDIX Extension: 1111



   Calling Party Number to INTUITY AUDIX? n
                            LWC Reception: none
```

Switch-Node 3 Administration

This section shows the screens for the administration of Switch Node 3.

DS1 Circuit Packs

```
add ds1 1a11                                                Page   1 of   2
                                DS1 CIRCUIT PACK

            Location: 01a11                        Name: Switch 3 to Switch 1
            Bit Rate: 1.544                 Line Coding: b8zs
   Line Compensation: 1                    Framing Mode: esf
      Signaling Mode: isdn-pri



Interface Companding: mulaw
           Idle Code: 11111111
             DMI-BOS? n




      Slip Detection? n                 Near-end CSU Type: other
```

Dial Plan Analysis

```
change dialplan analysis                                 Page   1 of   3   SPE A
                             DIAL PLAN ANALYSIS TABLE
                                                            Percent Full:    9


      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
      String  Length Type      String  Length Type      String  Length Type
        0       1    attd
        1       4    dac
        2       4    ext
        3       4    ext
        4       4    ext
        7       3    dac
        8       1    fac
        9       1    fac
        *       3    dac
        #       3    dac
```

Dial Plan Parameters

```
change dialplan parameters                                         Page 1 of 1
                              DIAL PLAN PARAMETERS

      AAR/ARS Internal Call Prefix: ________           Local Node Number:  3_
 AAR/ARS Internal Call Total Length: ___                 ETA Node Number:  __
                                                       ETA Routing Pattern:  __
             UDP Extension Search Order: local-extensions-first
         6-Digit Extension Display Format: xx.xx.xx
         7-Digit Extension Display Format: xxx.xxxx
```

Synchronization Plan

```
change synchronization                                            Page 1 of X
                                SYNCHRONIZATION PLAN
                    SYNCHRONIZATION SOURCE (circuit pack location)
                          Stratum: 4
                     Port Network: 1
                          Primary: 1a11_      Secondary: ___

        Location   Name      Slip   Type     Location   Name         Slip  Type
        01a11 DCS SW1_______   y    UDS1-BD  _____  _______________   _    ___
        _____ ______________   _             _____  _______________   _    ___
        _____ _______________  _             _____  _______________   _    ___
        _____ _______________  _             _____  _______________   _    ___
        _____ _______________  _             _____  _______________   _    ___
        _____ _______________  _             _____  _______________   _    ___
        _____ _______________  _             _____  _______________   _    ___
        _____ _______________  _             _____  _______________   _    ___
NOTE: TN722B & TN464B DS1 sources result in stratum 4, type II synchronization
```

Signaling Group

Page 1

```
add signaling-group next                                          Page 1 of 5

                           SIGNALING GROUP
Group Number: 1
                    Associated Signaling? y          Max number of NCA TSC: 5_
                       Primary D-Channel: 01a1124     Max number of CA TSC: 23
                                                    Trunk Group for NCA TSC: 13
       Trunk Group for Channel Selection: ___
          Supplementary Service Protocol: a__
```

Page 2 – Administered NCA TSC Assignment

```
add signaling-group next                                      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:   3901_     y     permanent   1901___________  dcs____  _______     1_
  2:   3902_     y     permanent   1902___________  dcs____  _______     2_
  3:   3903_     y     permanent   1903___________  dcs      _______     4_
  4:   3904_     y     permanent   1904___________  audix    _______     1_
```

Trunk Groups

Group 13 (ISDN-PRI) — page

```
add trunk-group 13                                          Page   1 of  10
                                TRUNK GROUP

Group Number: 13                   Group Type: tie          CDR Reports: y
  Group Name: DCS data to node 2 - TG22   COR: 1        TN: 1        TAC: 722
   Direction: two-way       Outgoing Display? n    Trunk Signaling Type:
 Dial Access? y               Busy Threshold: 99          Night Service:
Queue Length: 0                                    Incoming Destination:
   Comm Type: data                 Auth Code? n
                                 Trunk Flash? n

TRUNK PARAMETERS
   Trunk Type (in/out): wink/wink       Incoming Rotary Timeout(sec): 5
    Outgoing Dial Type: tone                       Incoming Dial Type: tone
                                              Disconnect Timing(msec): 500
       Digit Treatment:                                        Digits: __
                                                    Sig Bit Inversion: none
     Analog Loss Group:_______                     Digital Loss Group:
    Incoming Dial Tone? y

 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0            Receive Answer Supervision? y
```

Group 13 — page 2

```
add trunk-group 13                                                Page   2 of
10
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member: 1
                                      Send Name: n      Send Calling Number: n
            Used for DCS? y  PBX ID: 1
   Suppress # Outpulsing? n                                DCS Signaling: d-chan
 Outgoing Channel ID Encoding: exclusive     UUI IE Treatment: service-provider



                                                     Send Connected Number: n



               Send UCID? n
 Send Codeset 6/7 LAI IE? y
```

Group 13 — member assignments

```
display trunk-group 13                                            Page   4 of  10
                                TRUNK GROUP
                                      Administered Members (min/max):   0/0
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   0

       Port    Code Sfx Name         Night           Sig Grp
  1: 01a1101  TN464  e                                  1
  2: 01a1102  TN464  e                                  1
  3: 01a1103  TN464  e                                  1
  4: 01a1104  TN464  e                                  1
  5:
  6:
```

Uniform Dial Plan

NOTE:
Uniform Dial Plan administration changes considerably with R11. Now you use the Uniform Dial Plan Table to administer UDP. See the *Administrator's Guide for Avaya MultiVantage Software* for more information.

```
change uniform-dialplan 0                                               Page 1 of 2
                         UNIFORM DIAL PLAN TABLE
                                                       Percent Full:   2

Matching             Insert          Node     Matching          Insert          Node
Pattern   Len Del    Digits Net Conv Num      Pattern   Len Del Digits Net Conv Num
2____      4   0     817_   aar  n    __      _____     __  __   ___   __   __  __
4____      5   1     334_   aar  n    __      _____     __  __   ___   __   __  __
43659      5   1     928_   aar  y    __      _____     __  __   ___   __   __  __
623__      3   3     5380   aar  n    __      _____     __  __   ___   __   __  __
73012      5   1     ____   enp  n    31      _____     __  __   ___   __   __  __
74100      5   0     81__   ars  y    __      _____     __  __   ___   __   __  __
8____      5   0     ____   ext  n    __      _____     __  __   ___   __   __  __
911__      3   0     ____   ars  n    __      _____     __  __   ___   __   __  __
_____      _   _     ____   ___  _    __      _____     __  __   ___   __   __  __
_____      _   _     ____   ___  _    __      _____     __  __   ___   __   __  __
_____      _   _     ____   ___  _    __      _____     __  __   ___   __   __  __
_____      _   _     ____   ___  _    __      _____     __  __   ___   __   __  __
_____      _   _     ____   ___  _    __      _____     __  __   ___   __   __  __
```

AAR Digit Analysis

```
Change aar analysis 1                                          Page   1 of   2
                            AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:    6

          Dialed           Total     Route    Call   Node   ANI
          String          Min  Max  Pattern   Type   Num    Reqd
           221            7    7     101      aar    1      n
           222            7    7     101      aar    2      n
           224            7    7     101      aar    4      n
```

Routing Patterns

```
add route-pattern 101
                           Pattern Number: 101

   Grp.  FRL NPA Pfx Hop Toll No. Del Inserted                            IXC
   No.           Mrk Lmt List Digits  Digits
 1:13     0                   3                                           user
 2:                                                                       user
 3:                                                                       user
 4:                                                                       user
 5:                                                                       user
 6:                                                                       user

     BCC VALUE  TSC CA-TSC    ITC  BCIE Service/Feature          Numbering LAR
    0 1 2 3 4 W     Request                                      Format
 1: y y y y y n  y  as needed both ept                                     none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

Hunt Group

page 1

```
add hunt-group 99                                               Page   1 of  10
                                  HUNT GROUP

            Group Number: 99                                ACD? n
              Group Name: Intuity 1                       Queue? n
         Group Extension: 3111                           Vector? n
              Group Type: ucd-mia                  Coverage Path:
                      TN: 1         Night Service Destination:
                     COR: 1                    MM Early Answer? n
           Security Code:
     ISDN Caller Display:
```

page 2

```
add hunt-group 99                                               Page   2 of  10
                                  HUNT GROUP

                            Message Center: rem-audix
                           AUDIX Extension: 1111



    Calling Party Number to INTUITY AUDIX? n
                             LWC Reception: none
```

Switch-Node 4 Administration

This section displays the screens for the administration of Switch Node 4.

Bus Bridge

```
change system-parameters maintenance                            Page   2 of   3

                   MAINTENANCE-RELATED SYSTEM PARAMETERS

MINIMUM MAINTENANCE THRESHOLDS ( Before Notification )
        TTRs: 4        CPTRs: 1        Call Classifier Ports: 0
        MMIs: 0          VCs: 0

TERMINATING TRUNK TRANSMISSION TEST ( Extension )
   Test Type 100:          Test Type 102:          Test Type 105:

ISDN MAINTENANCE
   ISDN-PRI Test Call Extension:         ISDN-BRI Service SPID:

DS1 MAINTENANCE
   DS0 Loop-Around Test Call Extension:


SPE OPTIONAL BOARDS
                     Packet Intf1? y     Packet Intf2? y
         Bus Bridge: 01a05_  Inter-Board Link Timeslots  Pt0: 6 Pt1: 1 Pt2: 1
```

DS1 Circuit Packs

page 1

```
add ds1 1a10                                                Page   1 of   2
                                DS1 CIRCUIT PACK

            Location: 01a10                         Name: Switch 4 to Switch 1
            Bit Rate: 1.544                  Line Coding: b8zs
   Line Compensation: 1                     Framing Mode: esf
      Signaling Mode: robbed-bit



Interface Companding: mulaw
           Idle Code: 11111111



      Slip Detection? n                 Near-end CSU Type: other
```

Dial Plan Analysis

```
change dialplan analysis                                     Page   1 of   3   SPE A
                                   DIAL PLAN ANALYSIS TABLE
                                                                Percent Full:    9

      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
      String  Length Type      String  Length Type      String  Length Type
        0       1    attd
        1       4    dac
        2       4    ext
        3       4    ext
        4       4    ext
        7       3    dac
        8       1    fac
        9       1    fac
        *       3    dac
        #       3    dac
```

Dial Plan Parameters

```
change dialplan parameters                                                Page 1 of 1
                                    DIAL PLAN PARAMETERS

      AAR/ARS Internal Call Prefix: ________               Local Node Number:  4
 AAR/ARS Internal Call Total Length: ___                     ETA Node Number:  __
                                                           ETA Routing Pattern:  __
             UDP Extension Search Order: local-extensions-first
         6-Digit Extension Display Format: xx.xx.xx
         7-Digit Extension Display Format: xxx.xxxx
```

Synchronization Plan

```
change synchronization                                                Page 1 of X

                                       SYNCHRONIZATION PLAN
                          SYNCHRONIZATION SOURCE (circuit pack location)
                                Stratum: 4
                           Port Network: 1
                                Primary: 01a10       Secondary: ___

       Location   Name      Slip   Type      Location   Name          Slip  Type
       01a10 ______________   y    UDS1-BD   _____  _______________   _     ___
       _____ ______________   _              _____  _______________   _     ___
       _____ _______________  _              _____  _______________   _     ___
       _____ _______________  _              _____  _______________   _     ___
       _____ _______________  _              _____  _______________   _     ___
       _____ _______________  _              _____  _______________   _     ___
       _____ _______________  _              _____  _______________   _     ___
       _____ _______________  _              _____  _______________   _     ___
NOTE: TN722B & TN464B DS1 sources result in stratum 4, type II synchronization
```

Trunk Groups

Group 14 — page 1

```
add trunk-group 14                                            Page   1 of  10
                                TRUNK GROUP

Group Number: 14                  Group Type: tie           CDR Reports: y
  Group Name: Node 4 to Node 1 - TG14     COR: 1        TN: 1        TAC: 714
   Direction: two-way        Outgoing Display? n    Trunk Signaling Type:
 Dial Access? y                Busy Threshold: 99          Night Service:
Queue Length: 0                                     Incoming Destination:
   Comm Type: voice                  Auth Code? n
                                   Trunk Flash? n
         BCC: 0
TRUNK PARAMETERS
   Trunk Type (in/out): wink/wink        Incoming Rotary Timeout(sec): 5
    Outgoing Dial Type: tone                       Incoming Dial Type: tone
      Wink Timer(msec): 300                  Disconnect Timing(msec): 500
       Digit Treatment:                                        Digits:
                                                   Sig Bit Inversion: none
     Connected to Toll? n      STT Loss: normal       DTT to DCO Loss: normal
    Incoming Dial Tone? y
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0            Receive Answer Supervision? y
```

Group 14 — Page 2

```
add trunk-group 14                                              Page   2 of  10
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n
                                 Glare Handling: none
            Used for DCS? y  PBX ID: 1
   Suppress # Outpulsing? n
                    Seize When Maintenance Busy: neither-end




Incoming Tone (DTMF) ANI: no                   Per Call CPN Blocking Code:
         Connected to CO? n                  Per Call CPN Unblocking Code:
```

Group 14 — member assignments

```
add trunk-group 14                                              Page   4 of  10
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/3
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   3

       Port    Code Sfx Name        Night           Mode      Type   Ans Delay
  1: 01a1001  TN767  e                                                 20
  2: 01a1002  TN767  e                                                 20
  3: 01a1003  TN767  e                                                 20
  4:
  5:
  6:
```

Group 24 (data to Switch Node 1) — page1

```
add trunk-group 24                                       Page   1 of  10
                                TRUNK GROUP

Group Number: 24                  Group Type: tie           CDR Reports: y
  Group Name: dcs data to Switch 1        COR: 1        TN: 1        TAC: 124
   Direction: two-way        Outgoing Display? n    Trunk Signaling Type:
 Dial Access? y              Busy Threshold: 99          Night Service:
Queue Length: 0                                     Incoming Destination:
   Comm Type: rbavd                     Auth Code? n
                                  Trunk Flash? n
         BCC: 0
TRUNK PARAMETERS
   Trunk Type (in/out): wink/wink        Incoming Rotary Timeout(sec): 5
    Outgoing Dial Type: tone                        Incoming Dial Type: tone
      Wink Timer(msec): 300                  Disconnect Timing(msec): 500
       Digit Treatment:                                         Digits:
                                                   Sig Bit Inversion: none
     Connected to Toll? n      STT Loss: normal       DTT to DCO Loss: normal
    Incoming Dial Tone? y
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0            Receive Answer Supervision? y
```

Group 24 — page 2

```
add trunk-group 24                                               Page   2 of  10
TRUNK FEATURES
          ACA Assignment? n             Measured: none
                                  Internal Alert? n        Maintenance Tests? y
                                Data Restriction? n
                                  Glare Handling: none
            Used for DCS? n
   Suppress # Outpulsing? n
                    Seize When Maintenance Busy: neither-end





Incoming Tone (DTMF) ANI: no                    Per Call CPN Blocking Code:
         Connected to CO? n                   Per Call CPN Unblocking Code:
```

Group 14 — member assignments

```
add trunk-group 24                                                   Page   5 of  10
                                  TRUNK GROUP
                                       Administered Members (min/max):   1/3
GROUP MEMBER ASSIGNMENTS                   Total Administered Members:   3

       Port    Code Sfx Name         Night           Mode      Type   Ans Delay
  1: 01a1023  TN767  f                                                   20
  2:
  3:
  4:
```

Uniform Dial Plan

NOTE:
Uniform Dial Plan administration changes considerably with R11. Now you use the Uniform Dial Plan Table to administer UDP. See the *Administrator's Guide for Avaya MultiVantage Software* for more information.

```
change uniform-dialplan 0                                                Page 1 of 2
                         UNIFORM DIAL PLAN TABLE
                                                          Percent Full:   2

Matching           Insert          Node     Matching          Insert          Node
Pattern   Len Del   Digits Net Conv Num      Pattern   Len Del Digits Net  Conv Num
2          4   0    817    aar  n    __      _____     __  __   ___   __   __   __
4          5   1    334    aar  n    __      _____     __  __   ___   __   __   __
43659      5   1    928    aar  y    __      _____     __  __   ___   __   __   __
623        3   3    5380   aar  n    __      _____     __  __   ___   __   __   __
73012      5   1    ____   enp  n    31      _____     __  __   ___   __   __   __
74100      5   0    81     ars  y    __      _____     __  __   ___   __   __   __
8          5   0    ____   ext  n    __      _____     __  __   ___   __   __   __
911        3   0    ____   ars  n    __      _____     __  __   ___   __   __   __
_____      _   _    ____   ___  _    __      _____     __  __   ___   __   __   __
_____      _   _    ____   ___  _    __      _____     __  __   ___   __   __   __
_____      _   _    ____   ___  _    __      _____     __  __   ___   __   __   __
_____      _   _    ____   ___  _    __      _____     __  __   ___   __   __   __
_____      _   _    ____   ___  _    __      _____     __  __   ___   __   __   __
```

AAR Digit Analysis

```
change aar analysis 1                                             Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:    6

          Dialed            Total      Route     Call   Node   ANI
          String           Min  Max   Pattern    Type   Num    Reqd
           221             7    7      101       aar    1      n
           222             7    7      101       aar    2      n
           223             7    7      101       aar    3      n
```

Routing Patterns

```
change route-pattern 101
                            Pattern Number: 101

   Grp.  FRL NPA Pfx Hop Toll No. Del Inserted                            IXC
   No.           Mrk Lmt List Digits  Digits
 1:14     0                   3                                         user
 2:                                                                     user
 3:                                                                     user
 4:                                                                     user
 5:                                                                     user
 6:                                                                     user
```

Node Names

page 1

```
change node-names audix-msa                                 Page   1 of   1
                          AUDIX-MSA NODE NAMES

      Audix Names   IP Address          MSA Names    IP Address
      intuity1      192.168.200.10      msa              .   .   .
```

page 2

```
change node-names ip                                            Page 1 of 5
                           IP NODE NAMES

   Name                 IP Address          Name                 IP Address
 1.ppp41____________   192.168.200.14_  17. _______________    ___.___.___.___
 2.ppp14____________   192.168.200.12_  18. _______________    ___.___.___.___
 3.CMS______________   192.168.201.10_  19. _______________    ___.___.___.___
 4.router___________   192.168.200.1__  20. _______________    ___.___.___.___
 5.Ethernet1________   192.168.200.11_  21. _______________    ___.___.___.___
:
16.
```

Data Modules

PPP data module

```
add data-module 4991
                                                                  Page 1 of x
                          DATA MODULE

    Data Extension: 4991            Name: _ppp on link 1 to switch 1___  BCC: 2
              Type: PPP              COS: 1
              Port: 01a0515          COR: 1
              Link: 1_                TN: 1
       Enable Link? n

         Node Name: ppp41_________
       Subnet Mask: 255.255.255.0__


Establish Connection: n

DESTINATION
            Digits: 7241994___________
         Node Name: ppp14__________
             CHAP? n
```

Processor Channel Assignments

```
change communications-interface processor-channels
                                                                     Page 1 of X
                  PROCESSOR CHANNEL ASSIGNMENT

Proc                Gtwy        Interface    Destination      Session    Mach
Chan Enable  Appl.  To  Mode  Link/Chan   Node        Port  Local/Remote  ID
  1:    n  audix__         s     1_ 5041_  intuity__  0____  4__   4_     1__
  :
 10:    n  mis____         s     1_ 5001__ cms______  0____  1_    1_     __
  :
 14:    n  dcs____         c     1_ 0_____ ppp14____  5003__ 14_   14_    1_
 15:    n  dcs____         c     1_ 0_____ ppp14____  5004__ 15_   15_    2_
 16:    n  dcs____         c     1_ 0_____ ppp14____  5005__ 16_   16_    3_
  :
```

IP Routing

```
add ip-route next                                              Page   1 of   1
                              IP ROUTING


     Route Number: 1
 Destination Node: audix _
          Gateway: ppp14
      C-LAN Board: 1a05
           Metric: 0
       Route Type: host
```

IP Routing

```
add ip-route next                                              Page   1 of   1
                              IP ROUTING


     Route Number: 2
 Destination Node: cms_____
          Gateway: ppp14___
      C-LAN Board: 1a05
           Metric: 0
       Route Type: host
```

Hunt Group

page 1

```
add hunt-group 99                                                Page   1 of  10
                                  HUNT GROUP

            Group Number: 99                                ACD? n
              Group Name: Intuity 1                       Queue? n
         Group Extension: 4111                           Vector? n
              Group Type: ucd-mia                 Coverage Path:
                      TN: 1          Night Service Destination:
                     COR: 1                    MM Early Answer? n
           Security Code:
     ISDN Caller Display:
```

page 2

```
add hunt-group 1                                                Page   2 of  10
                                  HUNT GROUP

                           Message Center: rem-audix
                          AUDIX Extension: 1111



    Calling Party Number to INTUITY AUDIX? n
                            LWC Reception: none
```

Intuity Translations for DCS AUDIX

This screen is administered on Intuity, not on the MultiVantage solution.

```
                         Switch Interface Administration

   Switch Link Type:_LAN_                     Switch:_MultiVantage_____
   Extension Length:_4___                    Country:_United States_
 Host Switch Number:_1___
       Audix Number:_1___


 Switch  IP Address/Host Name  TCP Port   Switch  IP Address/Host Name  TCP Port
 Number                                   Number

  1__   192.168.200.11_______  5002__     11____  ___.___.___.________   ______
  2__   192.168.200.11_______  5021__     12____  ___.___.___.________   ______
  3__   192.168.200.11_______  5031__     13____  ___.___.___.________   ______
  4__   192.168.200.14_______  5041__     14____  ___.___.___.________   ______
  5__   _____________________  ______     15____  ___.___.___.________   ______
  :
  10_   _____________________  ______     20____  ___.___.___.________   ______
```

CMS Administration

See your CMS documentation for administration procedures for the CMS server.

Troubleshooting

7

This chapter describes troubleshooting methods for the TN799 C-LAN, the TN2302 IP Media Processor and the UDS1 tie trunk circuit packs.

Troubleshooting the TN2302 and TN799 circuit packs

If your TN2302 IP Media Processor or TN799 C-LAN circuit pack is not working, try these basic procedures before contacting Avaya for assistance. The following table lists some common circuit pack error messages and solutions.

Error Message	Solution
“Invalid board location; please press HELP”	Inspect board location. The entered board location is invalid or does not contain a C-LAN (TN799) board.
“No resource administered for this region”	Enter correct resource type on ip-interfaces screen.
“This board is not an administered IP-Interface”	Inspect board location. The entered board location contains a C-LAN which has not been administered on the ip-interfaces screen.

H.323 Trunk Troubleshooting

Signaling group assignments

You can assign multiple H.323 trunk groups to a single signaling group. However, when H.323 trunk groups have different attributes, assign each H.323 trunk group to a separate signaling group. An H.323 signaling group directs all incoming calls to a single trunk group, regardless of how many trunk groups are assigned to that signaling group. This is specified in the field “Trunk Group for Channel Selection” on the H.323 signaling group screen.

In the example shown in Figure 4, two trunk groups are assigned to the same signaling group on each of two switches, A and B. Trunk groups A1 and B1 are set up to route calls over a private network. Trunk groups A2 and B2 are set up to route calls over the public network. The signaling group on switch B terminates all incoming calls on trunk group B1 as specified by the “Trunk Group for Channel Selection” field. Calls from switch A to switch B using trunk group A1 and the private NW are terminated on trunk group B1, as desired. However, calls from switch A to switch B using trunk group A2 and the public NW are also terminated on trunk group B1, not trunk group B2, which is the desired outcome.

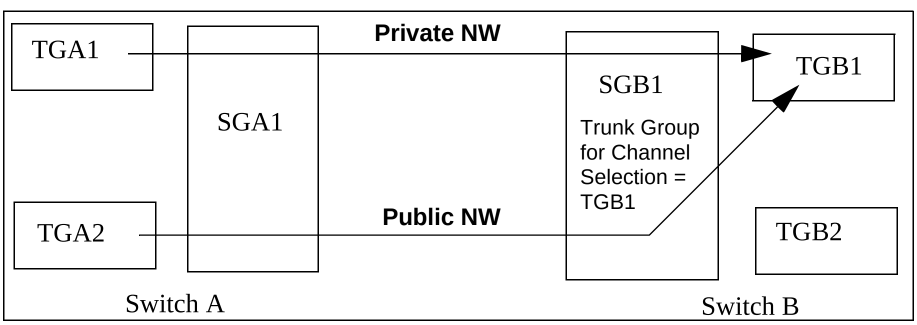


Figure 4. Shared signaling group

The solution to this problem is to set up a separate signaling group for each trunk group, as shown in the following figure. More generally, set up a separate signaling group for each set of trunk groups that have common attributes.

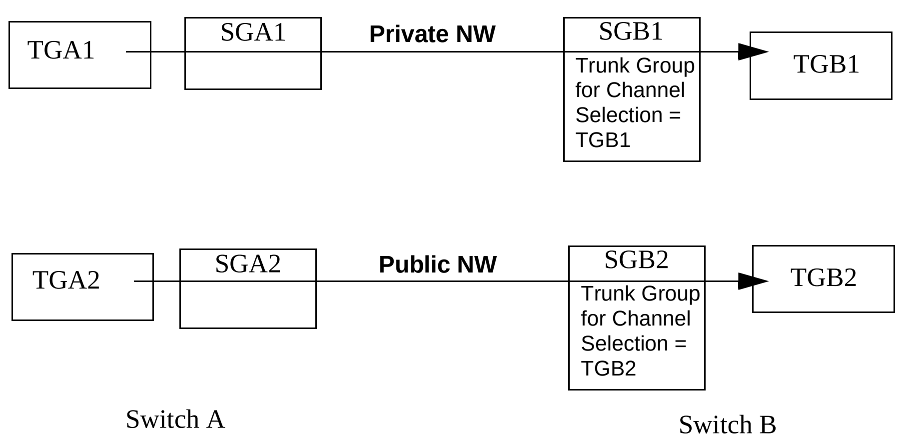


Figure 5. Separate signaling group

No MedPro resources available

If two switches are connected via an H.323 trunk and all MedPro resources are in use on the call-destination switch when a call is made, the call will fail even when a second preference is administered in the routing pattern on the source switch. This can be avoided by setting the first preference look ahead routing (LAR) to “next” in the routing pattern.

C-LAN sharing

Depending on the network configuration, a single C-LAN board can handle the signaling for multiple applications. For example, the call center Call Management System (CMS) typically uses a small portion of a C-LAN’s capacity so the same C-LAN can handle the signaling for other IP endpoints at the same time. There are many variables that affect the number of C-LAN circuit packs that you need for your network configuration. Contact your Avaya representative to discuss ways to accurately estimate the C-LAN resources you need.

Traffic congestion is potentially a problem when multiple IP Interfaces (such as C-LAN, IP Media Processor, PCs, CMS) share a network and some of the endpoints are heavily used. This problem can be minimized by using a switched network and assigning endpoints (such as CMS) to a separate LAN/WAN segment.

Troubleshooting Avaya IP telephones

If the Avaya IP Telephone installation or administration is not working, try these procedures before contacting your Avaya Service Center for assistance. The following table outlines some common IP Telephone troubleshooting symptoms.

Symptom	Solution
Unable to access IP Station screens	Make sure the IP Stations field in the Customer-Options Screen is set to **y**. If it is not enabled, you must obtain a new license file.
Port Field Display on the Station Screen reads “x”	The field defaults to **x** until a station registers for the first time. After the station has registered once, the port field shows the virtual LAN port address, even if the station unregisters.
IP Telephone not working	Use the status station command to see if the station is registered. The status should be “registered-authenticated”.

Troubleshooting problems with shuffling and hairpinning

Use the following procedures to maintain, review, and troubleshoot the status of stations, trunks, and IP network regions.

You can also access the Avaya support website at http://support.avaya.com. From there, you can search for additional troubleshooting information.

Reviewing a station's IP connection status

Use the status station command to determine the type of IP connection that is active.

1. Type **status station *extension*** to open the Call Control Signaling screen.
2. Move to the Call Control Signaling page.

```
status station 23484                                 Page 3  of 3
                    CALL CONTROL SIGNALING
                  Switch                    IP                                  IP
                 Port  Switch-end IP Addr: Port             Set-end IP  Addr: Port
  IP Signaling: 1C1417   xxx.xxx.xxx.xxx: nnnnn           xxx.xxx.xxx.xxx: nnnnn
         H.245:
     Node name:
Network Region:
                        AUDIO CHANNEL
         Switch                     IP                         IP
         Port   Switch-end IP Addr: Port        Set-end IP Addr:Port
        Audio: S0005 xxx.xxx.xxx.xxx: nnnnn xxx.xxx.xxx.xxx: nnnnn
     Node name:
Network Region:

  Audio Connection Type: ip direct
             Product ID: 1234SoftR2
 H.245 Tunneled inQ.931:
    Registration Status: authenticated
            MAC Address:
```

3. Review the following field:

Field	Value
Audio Connection Types	■ **ip-tdm -** connections from one endpoint through the TDM bus and back through the Media Processor ■ **ip-hairpin -** connection is between two endpoints that goes through the Media Processor but not through the TDM bus ■ **ip-direct** - connection goes directly between two endpoints without going through the Media Processor ■ **ip-idle** - the endpoint is idle and not connected

4. Exit the screen.

Reviewing a trunk's IP connection status

Use the status trunk command to determine the type of IP connection that is active.

1. Type **status trunk *group/member*** to open the Trunk Status screen.

```
status trunk 1/19                                         SPE B
                         TRUNK STATUS
Trunk Group/Member: 01/19                 Service State: in-service/active
          Port: T00123       Maintenance Busy? no
Signaling Group ID: 1                      CA-TSC state: not allowed
                             MM Conference ID: 8
                               MM Endpoint ID: 2

Connected Ports: 01B1431 01C1008
                 S00004

            Switch                   IP                      IP
            Port    Near-end IP Addr: Port   Far-end IP Addr: Port
     Q.931: 12B1217 xxx.xxx.xxx.xxx: nnnnn   xxx.xxx.xxx.xxx: nnnnn
     H.245: 12B1217 xxx.xxx.xxx.xxx: nnnnn   xxx.xxx.xxx.xxx: nnnnn
G.711-MU Audio:12B1108    xxx.xxx.xxx.xxx:  nnnnn     xxx.xxx.xxx.xxx: nnnnn

H.245 Tunneled in Q.931? no
Audio Connection Type: ip-tdm
```

2. Review the following field:

Field	Value
Audio Connection Types	■ **ip-tdm -** connections from one endpoint through the TDM bus and back through the Media Processor. For an IP-TDM call, the audio switch port field shows a port on a TN2302 Media Processor board. ■ **ip-hairpin -** IP connection is between two endpoints and goes through the Media Processor, but not through the TDM bus. For an IP-media processor-IP hairpin call, the audio switch port field shows a cabinet and slot, but not a port, on a TN2302 Media Processor board. ■ **ip-direct** - the IP-IP connection goes directly between two endpoints without going through the Media Processor. For an IP-IP direct call, the audio switch port field shows a virtual port number, for example, one starting with T. ■ **ip-idle -** IP endpoint is idle and not connected. If a trunk is IP-idle, the audio switch port field is blank.

3. Exit the screen.

Reviewing the IP network region status

Use the status ip-network-region command to determine if any of the IP network regions failed a ping test. If so, this indicates a connectivity failure between the network region you included in the command and the network region shown on the screen.

1. Type **status ip-network-region x** to open the Inter Network Region Connection Management screen.

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

Region                          (Group of 32)
          1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2
001-032   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ f _ _ _ _ _ _ _ _ _ _ _ _
033-064   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
065-096   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
097-128   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
129-160   _ _ 1 _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
161-192   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
193-224   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
225-256   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
257-288   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
289-320   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
321-352   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
353-384   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
385-416   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
417-448   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
449-480   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
481-500   _ _ _ _ _ _ _ _ _ _ _ _ _ _ _ _
```

2. Review the information on the screen.

   The values indicate that the two regions:

   - blank **-** are not administered
   - f **-** failed the maintenance ping test
   - 1 to 7 **-** passed the ping test. The number represents the preferred codec to be used between the two regions.

3. Exit the screen.

Displaying failed IP network region connections

Use the **display failed-ip-network-region** command to list the 100 network regions with highest number of broken connection paths. If a single network region has a large number of broken paths, the data equipment inside that region is probably the cause of the problem.

1. Type **display failed-ip-network-region** to open the first 100 worst network regions screen

```
Display failed-ip-network-region                     Page   1 of 1

          FIRST 100 WORST NETWORK REGIONS
      Network REGION: NUMBER OF BROKEN PATHS
_5:9_ _4:5_ _1:2_ __:__ __:__ __:__ __:__ __:__
__:__ __:__ __:__ __:__ __:__ __:__ __:__ __:__
__:__ __:__ __:__ __:__ __:__ __:__ __:__ __:__
__:__ __:__ __:__ __:__ __:__ __:__ __:__ __:__
__:__ __:__ __:__ __:__ __:__ __:__ __:__ __:__
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__:__ __:__ __:__ __:__ __:__ __:__ __:__ __:__
__:__ __:__ __:__ __:__ __:__ __:__ __:__ __:__
__:__ __:__ __:__ __:__
```

2. The network regions are ordered from worst to best.

   For example, in the pictured screen, region 5 has 9 broken paths (5:9) and region 4 has 5 broken paths (4:5).

3. Exit the screen.

Testing failed IP network regions

Use the **test failed-ip-network-region <#>** or **all** command to initiate a real-time ping test for all failed network-regions connections.

If there are no failed network-region connections, the network region connection warning alarm is cleared.

1. Type **test failed-ip-network-region <#>** or **all** to begin the test.
2. Test results screen appears at end of the test

```
                         TEST RESULTS

Port   Maintenance Name    Alt.Name   Test No.   Result   Error Code

        NR-CONN                 XXX-YYY      ZZZ          PASS/FAIL/ABORT
```

3. Review the test results.
   - NR-CONN represents the Maintenance Object Name for this test
   - XXX-YYY represents the pair of failed network regions being tested
   - ZZZ represents the test number
4. Exit the screen.

Considerations

Consider the following conditions when using hairpinning and shuffling.

Table 7. Considerations with hairpinning and shuffling

Condition	Solution
Audio Hairpin Connections come undone	The switch may undo hairpinning of audio connections, if a third party is conferenced into the existing two-party call, or when the switch wants to insert a tone or announcement into the connection, or for many other reasons. See Feature to Feature Interactions in this section for a complete list of situations that would cause the switch to undo a hairpin connection.
Volume is too quiet after a hairpin	The enduser using the Avaya endpoint would not have to adjust the volume control while the enduser using a non-Avaya endpoint may need to adjust the audio volume after the audio hairpinning is done.
Audio Shuffling Connections	The audio shuffling may cause a disruption in the media exchange for a duration of approximately 200ms. The disruption may be longer for an inter-network region call or a call traversing multiple switches. For a call involving a H.323 trunk as one of the endpoints, the administered value of the field “direct ip-ip audio connections” on the signaling group associated with that trunk determines the peer PBX’s Media Processor capability to handle shuffling. For a call traversing through multiple switches, the shuffling process may continue either leading to a full shuffle or a partial shuffle. For a normal point-to-point call between two IP terminals, the process can begin as soon as the terminating end answers the call. The call may undergo direct ip-ip audio connection or TDM connection based on user actions and feature interactions.

Continued on next page

Table 7. Considerations with hairpinning and shuffling *(Continued)*

Condition	Solution
The yellow LED on Media Processor board remains lit	As long as a TN2302 Media Processor board is hairpinning calls, its yellow LED will be lit. There is no simple way to identify all of the extension numbers that are hairpinning through a particular TN2302 Media Processor board. It is possible to determine which TN2302 Media Processor board a particular extension is using for hairpinning, by looking at the audio port field on the status station screen. A hairpinned call will show there as using a TN2302 Media Processor board slot, but not which TN2302 port.
TTD equipment is not sending or receiving tones accurately	If Teletype for the Deaf (TTD) equipment is to communicate over H.323 trunks, the system administrator should ensure that G.711 codecs are the primary codec choice for those trunks. This will ensure that the TTD tones are accurately sent through the connection.
Audio quality degrades	Audio quality may suffer if a call is subjected to a series of compressions of different types (some degradation is observed even if the same codec is used multiple times). If hairpinning or shuffling cannot be invoked, then maximum use of a G.711 codec should be encouraged to avoid multiple codec steps.
Switch ends IP audio channel	When and ip-media processor-ip hairpin or ip-ip direct call disconnects, if any set remains off-hook, the switch will connect to the set the appropriate tone as administered by the Station Tone Forward Disconnect field on the Feature-Related System Parameters screen. If that administered value is not silence, the switch will reconnect the audio path of such sets back to a TN2302 Media Processor port and the TDM bus, if an audio channel is available in the same network region. If that administered value is silence, the switch will end the IP audio channel.
Station cannot hairpin	If a 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 connection?” fields, the resulting station will not be allowed to hairpin. If a 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, the resulting station will not be allowed to hairpin.

Continued on next page

Table 7. Considerations with hairpinning and shuffling ***(Continued)***

Condition	Solution
User experiences one-way audio as soon as the far end connects	If an endpoint is not capable of shuffling, and also is not capable of signaling that limitation during registration, but the system administrator administers that endpoint to allow shuffling, the endpoint user will notice that two-party calls to other IP endpoints that are also administered as capable of shuffling will have one-way audio as soon as the far end answers the call, and similarly for calls from such endpoints.
Service Observer experiences break in speech path	If a call center agent is active on a two-party ip-ip direct call, and a call center supervisor chooses to service observe into the call, the agent would likely notice the 200ms break in the speech path while the call is being shuffled back to an ip-tdm-ip call. Stations that may be service observed should be administered to block shuffling.
LAN endpoint cannot be administered to allow shuffling	If a LAN endpoint is administered for permanent audio service link operation, the endpoint can not be administered to shuffle audio connections.
Calls are dropped during Busyout and Release	Busying out the TN2302 Media Processor board will drop all calls using the board in any manner. Note that calls carried by ip-ip direct audio connections are not using a TN2302 Media Processor board. Busying out ports 1-8 on the TN2302 Media Processor will drop all ip-tdm-ip hairpinned call and prevent future such calls on that port until the port is released, but not drop ip-media processor-ip hairpinned calls. Busying out a C-LAN board will cause the sets registered through that C-LAN to lose their registrations. If the sets are active on TDM connected or hairpinned calls, the calls will drop. Busying out a C-LAN board that is carrying signaling for tandem trunks causes all calls carried over those trunks to drop. What happens to calls carried by direct ip-ip audio connections when the corresponding C-LAN board is busied out depends on the endpoints involved in the call. Whether an endpoint will drop the call when it loses its registration depends on type of endpoint. In either case, the switch will not attempt to send new calls to unregistered sets.

IP Serviceability tools

IP Serviceability consists of switch-resident tools for troubleshooting communication problems between two IP endpoints within a system.

With the switch-resident tools, local and remote service personnel can troubleshoot, diagnose, and repair IP network-related problems without additional troubleshooting equipment.

This feature is a collection of report commands that monitor the TN799 (C-LAN) and TN2302 circuit packs. The following table lists these commands.

Table 8. IP Serviceability commands

Description	Command
Packet Loss/Jitter Size Report This information is not available for shuffled calls.	**status station** and **status trunk xx/yy**
IP Denial Events Log	**display events** event type: **denial**
Netstat ARP Report Address Resolution Protocol (ARP) is a low-level protocol that “maps” IP addresses to their corresponding MAC addresses	**netstart arp**
Refresh route tables	**refresh route-table**
IP Denial Events Log	**display events**
Trace Route Results	**trace-route node-name**
Trace Route Results	**trace-route ip-address**
Ping Results	**ping ip-address**

Fixing problems / troubleshooting strategy

The following table lists some of the most common IP problems, their cause and possible solutions.

Table 9. IP troubleshooting guide

Symptom	Possible cause	Solution
Noisy connection that eventually breaks up, creating gaps in the conversation and making speech unintelligible.	Packet loss Jitter	Depending on the “side” of the switch, use: status station status trunk xx/yy
IP service not responsive or cannot make connection	IP connection problem	Use the **display events** command with event type: **denial** to look for denied IP events.

Diagnostic and troubleshooting commands

The following table lists diagnostic and troubleshooting commands and their use.

<table>
<tr><th>Command</th><th>Description</th><th>Use</th></tr>
<tr><td>display events

event type: denial</td><td>This command creates a list of each time an IP event is denied, rejected, or some service is not provided. Each log entry consists of
a 25-character string that describes the problem
an identification number that uniquely implicates a specific piece of code
another numeric string describing the denial</td><td>Accumulate data to track denied or rejected IP service.</td></tr>
<tr><td>status station</td><td rowspan="2">This command generates a report that shows:
the number of packets that are either lost or corrupted
amount of jitter on the connection. In this instance, jitter is the variability in the amount of time, in milliseconds, that packets are received over the network. When jitter increases, the user experiences a noisy connection, delays, and a general loss of quality, making speech unintelligible.</td><td>Diagnose for station-side IP problems, including the IP phone.</td></tr>
<tr><td>status trunk xx/yy</td><td>Diagnose for trunk-side IP problem.</td></tr>
</table>

Troubleshooting IP Softphone

Codecs used with NetMeeting

Voice quality for the road-warrior application of IP softphone will vary depending on several factors. Poor voice quality can be caused by the use of the high-compression codecs (G.723 or G.729) in situations where the low-compression codec (G.711) should be used. This can happen unexpectedly when using NetMeeting. When the G.711 is set as the preferred codec on the switch, NetMeeting may fail to use it. Since there is no way to monitor which codec is being used, the only way to determine that this is the problem is to disable all but the G.711 codecs. If calls no longer work, it can be concluded that NetMeeting is failing to use the G.711.

The solution to this problem is to upgrade to the latest release of SoftPhone, which no longer uses NetMeeting.

Telecommuter use of phone lines

The telecommuter application of the IP Softphone requires the use of two phone lines: one for the IP connection to MultiVantage, which is used for softphone registration and call signaling, and the other for a PSTN connection, which MultiVantage uses as a callback number to establish the voice path. How you allocate your phone lines to these two functions may make a difference.

For example, assume that you have telephone services provided by the local phone company, such as voice mail, associated with one of your lines and not the other. In this case, you should use the line with the services to make the initial IP connection to register the softphone and use the line without the services as the POTS callback for the voice path. Otherwise, there could be undesirable interactions between the softphone and the local services. For example, if your telecommuter application is registered and you were using your POTS callback line for a personal call when a business associate dialed your work extension, the business associate would hear your home voice mail message.

iClarity audio level adjustments

When your system uses iClarity, and you have trouble hearing the audio on calls, you can use the Avaya IP Softphone Audio Control toolbar and the Audio Status dialog box to check microphone volume and channel power (speakers and headsets) while you are on an active call. You can also use the tools menu to check bandwidth settings and gain. You can run the Tuning Wizard to retrain Avaya iClarity IP Audio to the level of background noise at your location. See your IP Softphone online help for more information.

You can access the Avaya support website at http://support.avaya.com. From there, you can search for additional information, including:

- Recommended Headsets for IP Softphone and IP Agent
- Recommended soundcards for IP Softphone and IP Agent
- USB Headset information
- Avaya IP voice quality Network requirements, including VPN and NAT information

NetMeeting drops unanswered calls

For calls made to a NetMeeting softphone, if the call is not answered or if coverage is not administered, after about 5 rings NetMeeting drops the call and the station stops ringing but the caller continues to hear ringback.

The solution to this problem is to upgrade to the latest release of SoftPhone, which no longer uses NetMeeting.

NetMeeting ignores out-of-band tones

NetMeeting ignores any H.323 digits received out-of-band, so it never hears DTMF from the switch, which always strips DTMF from the audio path and plays it out-of-band with H.323. For example, if you are on a call on a NetMeeting softphone and the calling party presses a number or character on their keypad, you will not hear the tone.

The solution to this problem is to upgrade to the latest release of SoftPhone, which no longer uses NetMeeting.

Private Networking

A

This appendix 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 features for DCS, including "Private Network Access", and "Extension Number Portability".
- "Centralized Attendant Service" contains a description of CAS and features, considerations, and feature interactions.
- "Extended Trunk Access" contains description, administration, and feature interactions for ETA.
- "Inter-PBX Attendant Service" contains description, administration, and feature interactions for Inter-PBX Attendant Service.
- "ISDN Feature Plus" is a description of ISDN Plus networking capability.
- "Centralized Voice Mail Via Mode Code" is a brief description of Centralized Voice Mail with Mode codes.
- "Japan TTC Q931-a Private Networking Protocols" is a brief description of Japan TTC private networking protocols.

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 MultiVantage features. (Feature transparency means that features are available to all users on DCS regardless of the switch location.)

Configuring a DCS network is a complex process that involves 4 major steps:

- Planning your DCS network
- Connecting the physical equipment in the network
- Administering the physical layer (hardware connections)
- Administering the link layer to create a DCS

Prerequisite DCS Features

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 MultiVantage Software.*

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.

Required administration for Private Network Access

Screen	Field
Trunk Groups Access APLT ISDN-BRI ISDN-PRI Tandem	■ All
Class of Restriction	■ Advanced Private Line Termination
Feature Access Code (FAC)	■ Automatic Alternate Routing Access Code
Dialplan Analysis form	■ All

Screen	Field
AAR and ARS Digit Conversion Table	■ All
Node Number Routing	■ All
Station	■ COR

- Trunk Group Screens — Set the **Group Type** field to **access**, **aplt**, **tandem**, **tie**, or **isdn** and the **Service Type** field to **access**, **tie**, or **tandem**. Complete COR digit treatment and common type fields for tie trunk groups associated with a private network.

Detailed description

Private networks can include:

- Common-control switching arrangement (CCSA)
- 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
- QSIG Trunks (for more information, see Chapter 5, "QSIG")
- IP Trunks

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 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.

In Italy, TGU/TGE/TGI 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.

Interactions

- Attendant Call Waiting

  Call Waiting is provided via 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 via TGU/TGE trunks. Attendant Intrusion also is provided through DCS.

Extension Number Portability

Extension Number Portability (ENP) gives you the ability to assign any extension to any switch in an ENP subnetwork. Stations can be moved across switches while retaining the original extension number, as long as the switches are part of the ENP subnetwork. ENP is used in conjunction with Automatic Alternate Routing (AAR) and Uniform Dial Plan (UDP).

How to administer ENP

Screen	Field
AAR and ARS Digit Conversion Tables	■ All
Extension Number Portability Numbering Plan	■ All
Node Number Routing	■ All
Uniform Dialing Plan	■ Ext Code

- AAR Digit Conversion screen — 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.
- ENP Numbering Plan screen — 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 screen — Associate a route pattern with each node in the ENP subnetwork.
- Uniform Dialing Plan screen — 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).

Detailed description

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).

Interactions

- Distributed Communications System

  If you use DCS, the ENP node numbers must correspond to DCS node numbers.

Description of DCS

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 MultiVantage solution 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.

DCS Connectivity

ISDN/X.25 gateway

An Avaya MultiVantage solution 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.

It 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.

In summary, 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.

DCS Over ISDN-PRI D-channel

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+) enhances DCS by allowing access to the public network for DCS connectivity between DCS switch nodes.

The ISDN-PRI B-channel is used for voice communications, and the ISDN-PRI D-channel transports DCS control information. 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. 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.

How to administer DCS Over ISDN-PRI D-channel

Screen	Field
Signaling Group	■ Max number of NCA TSC ■ Max number of CA TSC ■ Trunk Group for NCA TSC ■ Administered NCA TSC Assignment fields ■ Service/Feature ■ Inactivity Time-out (min)
ISDN TSC Gateway Channel Assignments	■ All
Trunk Group (ISDN-PRI)	■ Used for DCS Node Number DCS Signaling ■ NCA TSC Trunk Member
Route Pattern	■ TSC ■ CA TSC Request
Processor Channel Assignment	■ Application
Feature-Related System Parameters	■ Record TSCs for CDR

NOTE:
There are several differences in administration between switches. For example, PRI is translated a little differently in 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 are used in combination, translations are also different.

Detailed description

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 (CA-TSC)
- Non-Call Associated (NCA-TSC)

CA-TSC

A 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 MultiVantage solution, this type of TSC is used only for DCS features on ISDN-PRI Signaling Groups administered with Supplementary Service Protocol a.

NCA-TSC

An NCA-TSC is a connection not related with any ISDN B-channel connections. MultiVantage 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 via 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.

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 via the external modem.

DCS Features

Once you have connected and set up your DCS network, you can provide the following features across the network:

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.

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.

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.

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.

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.

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 multiappearance voice-terminal users to make test calls to voice terminals and trunk groups that are located at other nodes in the DCS.

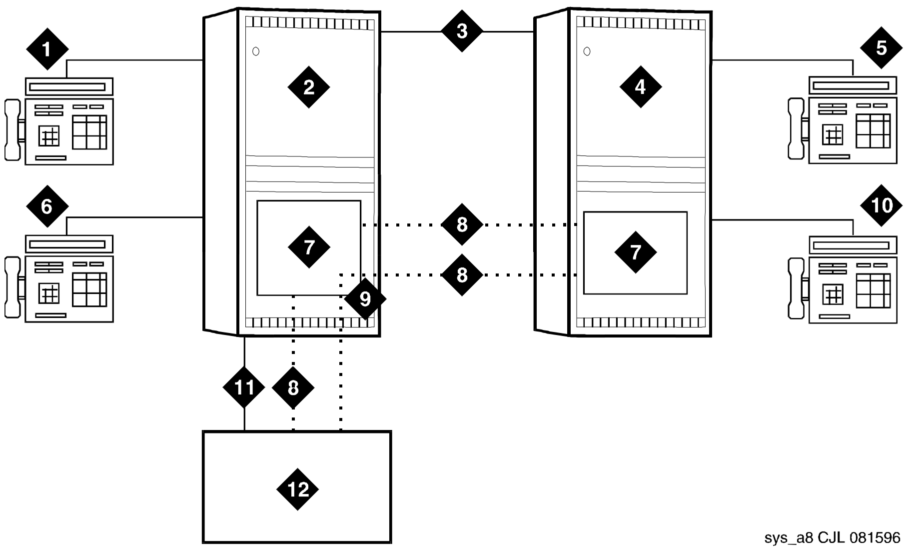
To use this feature, you must have a DCS Trunk Group between the local and remote switches, and the trunks in that trunk group cannot insert digits on incoming calls. If you need digit insertion on these trunks, it should be added on the outgoing trunk based on the dialed digits.

Multiappearance 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.

Call Coverage

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.

The following figure shows an example of DCS Call Coverage.



1. **Station A**
2. **System A - DEFINITY ECS**
3. **DCS Tie Trunk Groups**
4. **System B - DEFINITY ECS**
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**

Figure 6. DCS Call Coverage

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.

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.

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.

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.

Leave Word Calling

LWC transparency in a DCS configuration allows messages from a MultiVantage switch to another node, depending on the storage capability of the remote node.

Multiappearance Conference/ Transfer

DCS Multiappearance 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.

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.

DCS with Rerouting

DCS with Rerouting allows a call's connection between two MultiVantage solutions 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 MultiVantage solution.
- 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 Reroute perform the following administration:

1. On the System-Parameters Coverage screen, enter **n** in the Maintain SBA at Principal field. DCS with rerouting can only occur if you <u>do not</u> need to maintain a simulated bridged appearance at the principal.
2. On 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**.

Beginning in Release 6.1, users invoked DCS with Rerouting by Call Transfer, Transfer out of AUDIX, and dial 0 out of AUDIX.

Feature 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 a MultiVantage solution 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 a MultiVantage solution 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 MultiVantage solution functions as an incoming gateway and rerouting occurs.
- If a conference call is transferred, rerouting will not occur.

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.

How to administer Enhanced DCS

Screen	Field
Feature-Related System Parameters	■ ITALIAN DCS PROTOCOL Italian Protocol Enabled ■ Apply Intercept Locally ■ Enforce PNT-to-PNT Restrictions

DCS feature considerations

Attendant

- 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.

Alphanumeric Display considerations

- 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 MultiVantage solutions.

Attendant Control of Trunk Group Access considerations

- 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.

Attendant Direct Trunk Group Selection considerations

- This feature is not available for trunk groups with 4-digit trunk access codes or for trunk members 100 through 999.

Attendant Display considerations

- CORs for a MultiVantage switch may 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.

Automatic Callback considerations

- 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.

DCS Over ISDN-PRI D-channel 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.

Enhanced DCS 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.

LWC considerations

- 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.

Trunk Group Busy/Warning Indication considerations

- 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 Interactions

Alphanumeric Display

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.

DCS Attendant Control of Trunk Group Access

- 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 via 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.

Attendant Display

- 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.

Automatic Callback

- 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.

Busy Verification

- 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

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.

Call Forwarding

- 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 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.

DCS Over ISDN-PRI D-channel

- 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 directly select an extension that has been previously administered as belonging to a administered NCA-TSC results in intercept tone being received.

- CDR

  CDR records both the status and the utilization 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 recovered from.

- 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 intercept tone being received.

Distinctive Ringing

- Distinctive Ringing

  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 Example DCS configurations (page 206).

Enhanced DCS

- Class of Restriction

  When a call goes to coverage, it is the called party's (not the covering party's) restrictions that are used.

LWC

- DCS Multi-appearance Conference/Transfer

  Activation of LWC is denied after a DCS call has been conferenced or transferred.

Multiappearance Conference/Transfer

- 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.

Trunk Group Busy/Warning Indication

- 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.

Example DCS configurations

The following two examples provide details for setting up two basic DCS networks. 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. For examples of TCP/IP signaling for DCS, see Chapters 2 and 3 in this book.

2-Node private network with AUDIX

Figure 7 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 via 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 via 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.

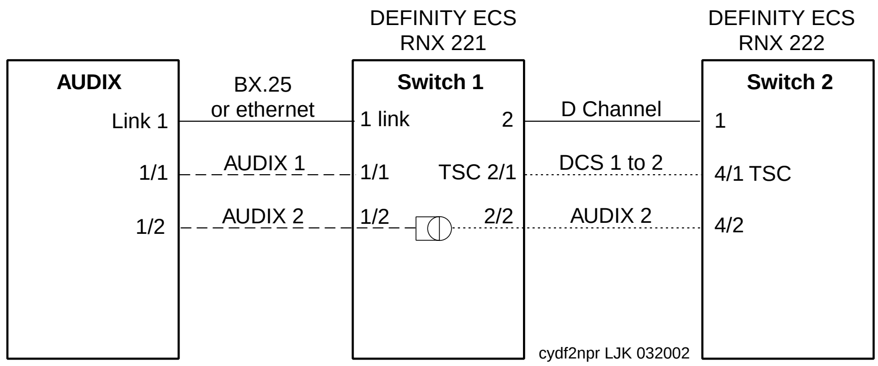


Figure 7. 2-Node private network

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.

Administration for switch 1

- Dial Plan Analysis screen

Dialed String	Total Length	Call Type
4	4	ext
5	4	ext

- Uniform Dial Plan screen

Ext Code	Type	Location Code
5xxx	UDPcode	222

- 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

- 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

Administration for switch 2

- Dial Plan Analysis screen

Dialed String	Total Length	Call Type
4	4	ext
5	4	ext

- Uniform Dial Plan screen

Ext Code	Type	Location Code
4xxx	UDPcode	221

- 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

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 8 shows a 3-node network.

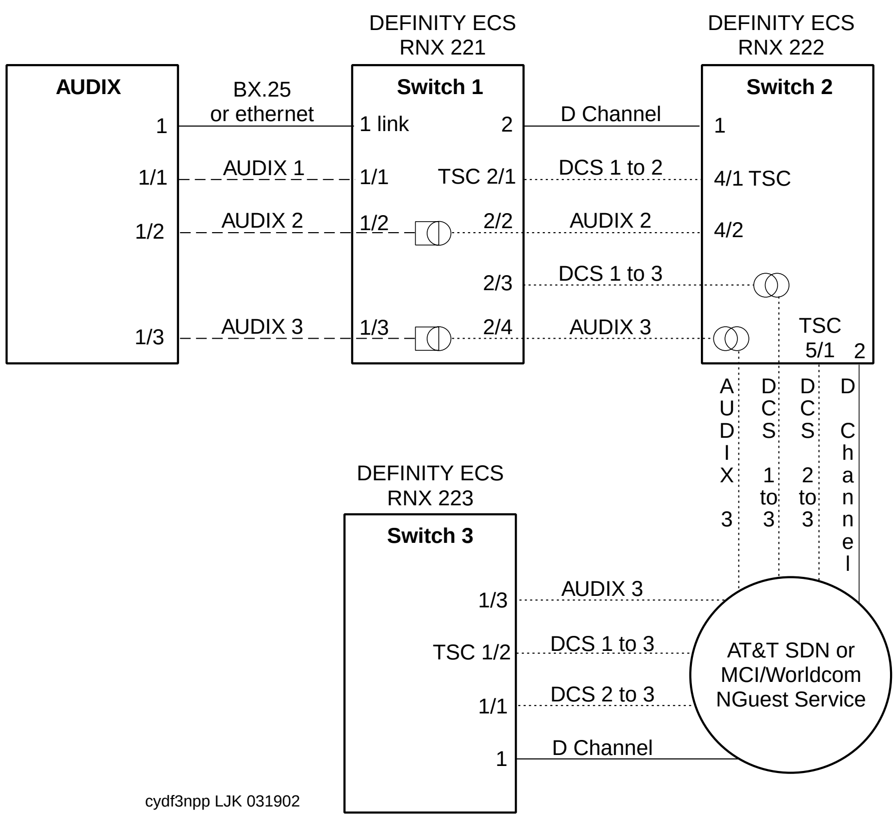


Figure 8. 3-Node public/private network

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.

Administration for 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	UDPcode	222
6xxx	UDPcode	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	-

Administration for 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	UDPcode	221
6xxx	UDPcode	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

Administration for 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	UDPcode	221
5xxx	UDPcode	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.

Centralized Attendant Service

Centralized Attendant Service (CAS) 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.

How to administer CAS

Screen	Field	Page
Attendant Console	■ Feature Button Assignments — cas-backup -trunk-name	5-38
Console-Parameters	■ CAS ■ RLT Trunk Group Number ■ CAS Back-Up Ext ■ Timed Reminder on Hold ■ Return Call Timeout (sec)	5-90
Station (multi-appearance)	■ Feature Button Assignments — cas-backup — flash — trunk name — night serv	6-30
Trunk Group (RLT)	■ All	7-238
Feature Access Code (FAC)	■ CAS Remote Hold Access Code	5-120

Detailed description

Each branch in a CAS has its own 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.

Each branch in a CAS network 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

A branch can connect to only one main.

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 feature and provide notification that backup service is in effect, 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 feature, 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 should 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.

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 multiappearance 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.

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 via 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 via Send All Calls to the backup extension's coverage path.
- Call Detail Recording

  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.

- DXS and 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 TAAS.

- Night Service — Trunk Answer from Any Station

  In a multiswitch DCS environment with CAS, the result of transferring incoming trunk calls via 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).

- Nonattendant 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 Any Station feature.

Extended Trunk Access

Use Extended Trunk Access (ETA) 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.

How to administer ETA

Screen	Field
Dial Plan Record (First Digit and Second Digit Tables)	■ 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.

Detailed description

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.

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.

Interactions

- Abbreviated Dialing

  Abbreviated Dialing calls are routed via ETA.

- Attendant

  Attendants calls are routed via 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 via 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

Inter-PBX Attendant Service (IAS) 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.

How to administer Inter-PBX Attendant Service

Screen	Field
Tie Trunk Group (Main)	■ Incoming Destination
Console Parameters (Branch)	■ IAS (Branch) ■ IAS Tie Trunk Group No. ■ IAS Att. Access Code
Tie trunk group (Branch)	■ All

Detailed description

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 MultiVantage solution 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.

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 via 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

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 for using the switched public network.

How to administer 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.

1. On the System-Parameters Customer-Options screen, verify that the:
   - **G3 Version** is **V7** or higher.
   - **ISDN Feature Plus** field is set to **y**.
2. On the same screen, 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, 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, to add a centralized AUDIX system, set the **Message Center** field to **fp-mwi**.

To start administration for Message Waiting Indication at the Message Center PBX:

1. On the Feature Related System Parameters screen, 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. Administer the Message Waiting Indication Subscriber Number Prefixes screen. To start the Calling Name feature:
4. On the ISDN-BRI or ISDN-PRI trunk group screen (whichever you are using), set the **Send Name** field to **y**.

Description

ISDN Feature Plus uses a MultiVantage 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 DID extensions. In addition to the general Feature Plus call handling, Feature Plus includes the following features:

- 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 assign transfer calls between parties across the public network is supported. 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 assign display a message waiting indication on a user’s voice terminal.

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 Feature Plus. This difference is due to the difference in 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, while 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 Chicago is a remote PBX:

- If Feature Plus is running between Dallas and Chicago and the Feature Plus extension in Chicago is 82000, the **Inserted Digits to form Complete Number** field administered in Dallas to get to Chicago must be 3035382000. The **Routing Digits (AAR/ARS Access Code)** field also needs to be filled in appropriately.
- If QSIG is running between Chicago 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.)

Interrogation

When performing an audit, the Served User switch sends a request towards the Message Center switch. As a Message Center PBX, the Avaya MultiVantage solution 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 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.

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 is 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 MultiVantage solution 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 MultiVantage 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 MultiVantage 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 Via Mode Code

You can use a single voice mail system to support multiple MultiVantage and Merlin Legend/Magix systems in a network via mode code. This capability is available for:

- DEFINITY ECS R8 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

Configuration requirements

Centralized voice mail via mode code requires the following:

- A MultiVantage solution 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:
To use Centralized Voice Mail via Mode Code, you're network must be in a hub/spoke configuration, with no more than ten DCS network nodes.

Configuration example

Figure 9 shows what a configuration of centralized voice mail via mode 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.

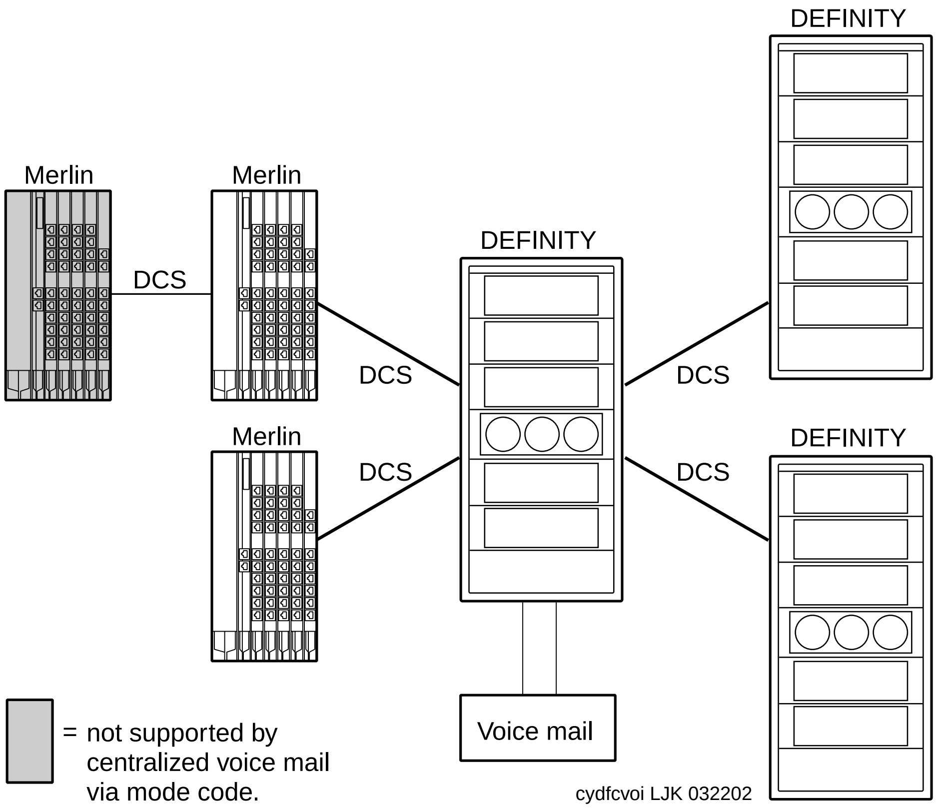


Figure 9. Centralized Voice Mail Via Mode Code Example Configuration

Feature Support

Features that are supported are:

- Calling party name/number sending/retrieval
- Message waiting light activation
- Remote coverage to voice mail
- Fax, as well as voice, mail

Features Not Supported

The following capabilities are not supported via 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

Setting Up Centralized Voice Mail Via Mode Code

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 your Intuity documentation.

1. Enter **display system-parameters customer-options** on the command line of your system administration screen.
2. On page 3, verify fields as follows:
   — **ISDN-PRI** field is **y**.
   — **Mode Code for Centralized Voice Mail?** field is **y**.
   — **Mode Code Interface?** field is **y**.
   — **UDP** field is **y**.
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 Typ**e 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.**
   — **Numbering 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 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 Code** 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 sign next** on the command line. Administer the signaling group (see example in Chapter 4 of this document).
14. For each node, enter **change isdn 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 MultiVantage node, enter **change sys params mode-code** on the 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 an** on the 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 command line.
24. Administer or verify the following:
    - — **01** contains the extension of the voice mail hunt group.

Japan TTC Q931-a Private Networking Protocols

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 modify other international standards as defined by ITU-T for use in Japan with additional national protocols to enhance operation for their customers.

Overview

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. MultiVantage provides connectivity into the Japanese private networking environment via two methods

- CAS (Channel Associated Signaling) — first developed in DEFINITY ECS R7 for the Japan market.
- ISDN (Integrated Services Digital Network) PRI (Primary Rate Interface) — TTC specific protocol supported on DEFINITY ECS R8 or newer systems. MultiVantage supports Basic Call with Number Identification services.

TTC Basic Call Setup with Number Identification Supplementary Service

MultiVantage allows the display of the calling party number to the called party. MultiVantage 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; 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 MultiVantage 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).

TTC Q931-a Protocols

The TTC defined private networking ISDN protocol is largely based upon ITU-T Q.931 protocol. MultiVantage 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.

Setting Up TTC Q931-a

Complete the following steps to set up TTC connections.

Steps

1. Verify that you have the appropriate circuit pack for integration
2. Enter “**change system-parameters customer-options**” on the command line of your system administration screen.
3. On page 1, verify that the G3 Version field is **V8** or later
4. On page 2, 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: (*This item must match between the local and receiving switches*)

   — Channel Numbering — **sequential** or **timeslot** (*This item must match between 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**

   Check for the following field entries On page 2

   — Disconnect Supervision — **y**

   — Numbering format — **public, private, unknown, unk-pvt**

   — Send Called/Busy/Connected Number — **y**

   — Sending Calling Number — **y**

   — Send Name — **n**

IP Routes

B

Using IP routes

On LANs that connect to other networks or subnetworks, Avaya recommends that you define a default gateway. See "Default gateway" 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 your MultiVantage Solution. 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.

IP routing screen

The following diagram shows the 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
```

To set up an IP route, enter the node names for the destination and the gateway, and enter the slot location of the C-LAN on the local switch. The destination and gateway node names and their associated IP addresses must be specified on the Node Name 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. (see "Network IP addresses" for a description of a network address)

Advanced IP routing

If you wanted the local C-LAN node to be able to communicate with the nodes on the 192.168.1.64 subnetwork and not with others, you could do the following:

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

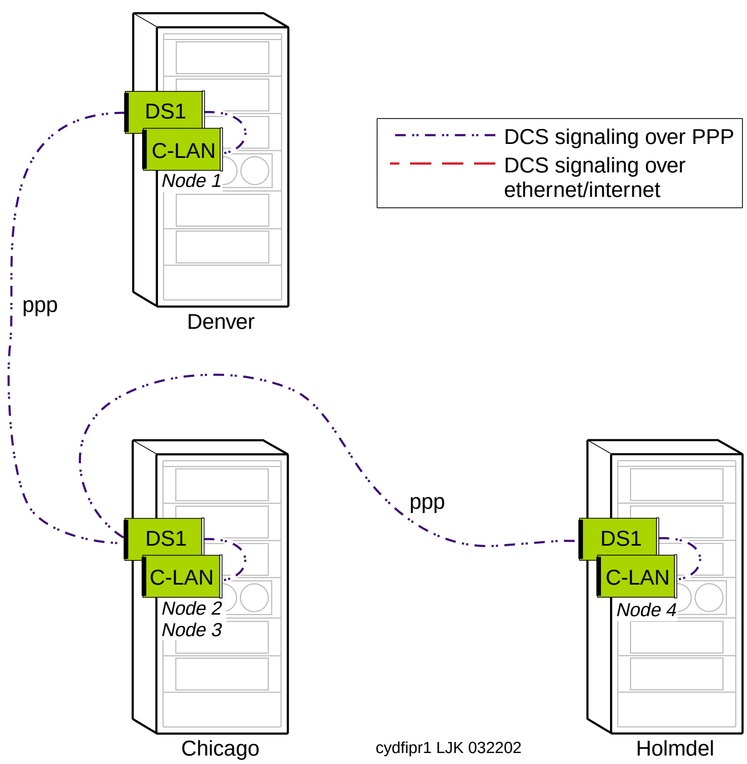
See the description of the subnet mask in "Subnetting" for more information on subnet addresses.

IP route example: PPP connections

The diagram below shows a DCS network with PPP signaling connections between systems Chicago and Denver, and between systems 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.



NOTE:
The IP routes between nodes for this example are listed in the following table. 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.

MultiVantage system	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

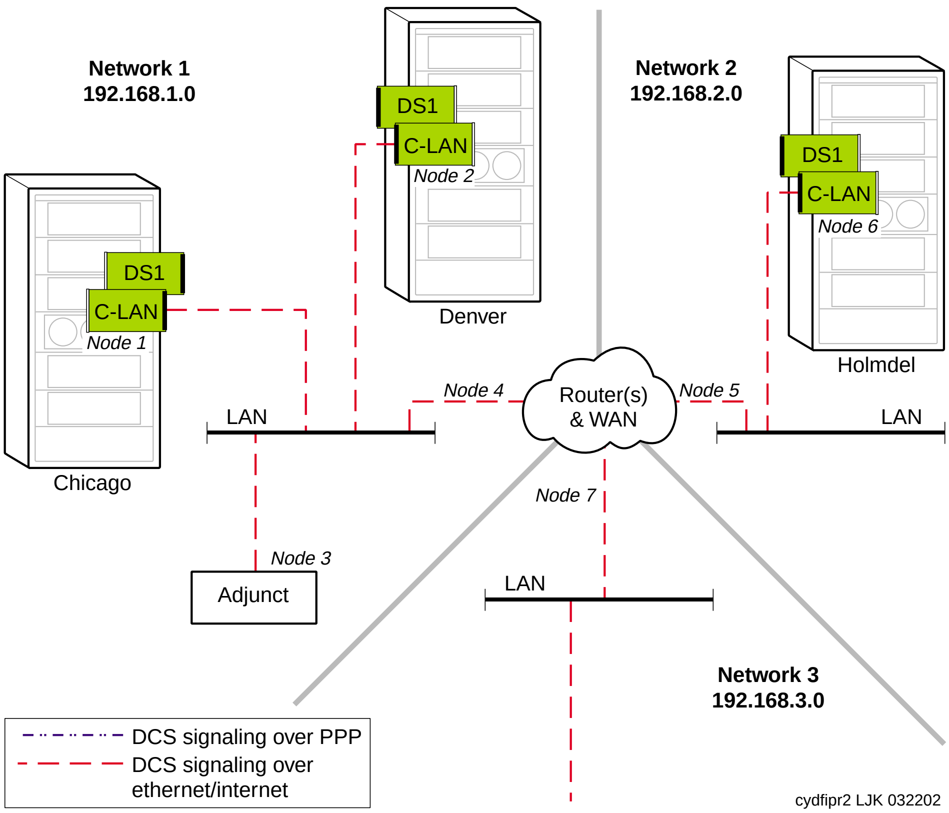
The diagram below shows two interconnected (sub)networks. There are three systems in a DCS network with a PPP signaling connection between systems in Chicago and Denver and an Ethernet signaling connection between the MultiVantage in Chicago and the adjunct. Chicago and Denver and the adjunct are on 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 via PPP and there are intermediate nodes between Denver & Holmdel.



The following table shows the IP routes needed if nodes 5 and 6 *are* not defined as default gateways for nodes 2 and 7.

Table 10. IP route examples

Avaya MultiVantage Solution	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 via 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 via 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.

Continued on next page

Table 10. IP route examples *(Continued)*

Avaya MultiVantage Solution	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.

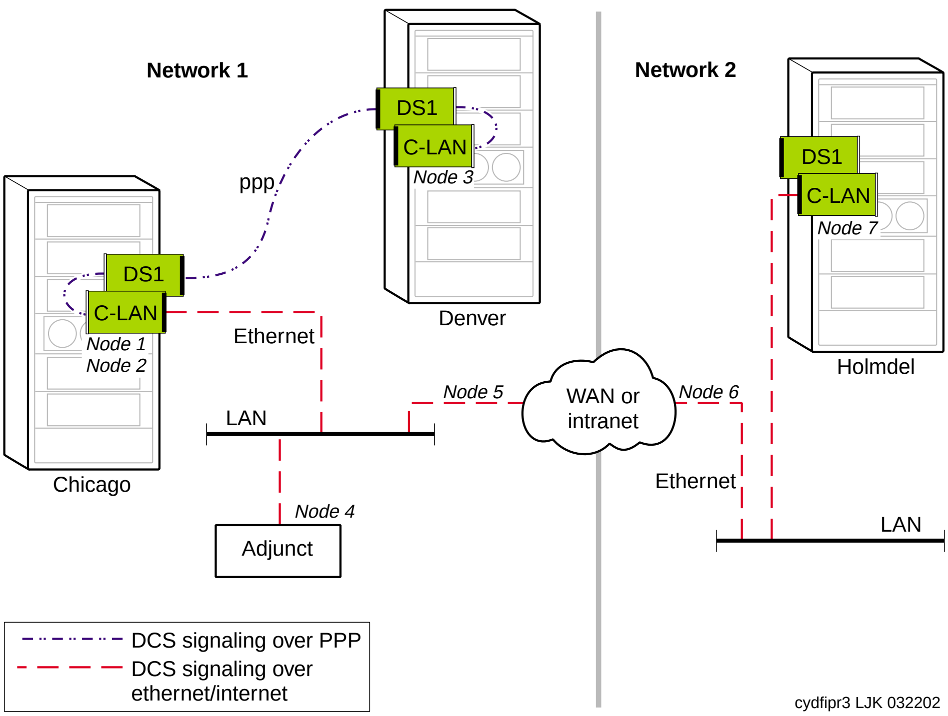
IP route example: Ethernet-only connections

The diagram below shows three interconnected (sub)networks. There are three systems in a DCS network with Ethernet signaling connections between them. MultiVantage systems 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.



The following table 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 11. IP route examples (if node 4 is not defined)

Avaya MultiVantage Solution	IP Node Connec-tions	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	netwo rk	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.

IP Addressing

C

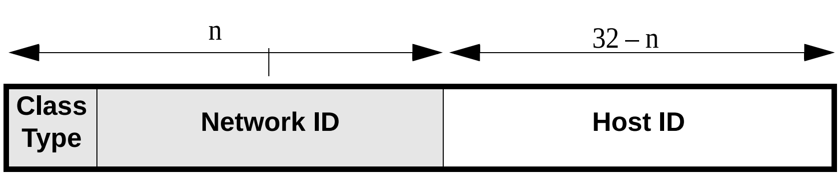
IP Addressing

This section describes IP addressing, subnetting, and routing.

Logical Addressing

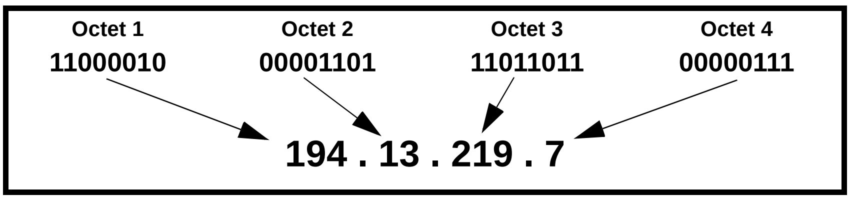
An IP address is a software-defined 32-bit binary number that identifies a network node. The IP address has two main parts -- the first n bits specify a “network ID” and the remaining 32 – n bits specify a “host ID.”

Format



Dotted Decimal notation

The 32-bit binary IP address is what the computer understands. For human use, the address is typically expressed in dotted decimal notation — the 32 bits are grouped into four 8-bit octets (bytes) and converted to decimal numbers separated by decimal points, as in the example below.



The eight binary bits in each octet can be combined to represent decimal numbers ranging from 0 to 255.

Conversion between binary and decimal

Conversion from binary to decimal notation is accomplished by adding the powers of 2 corresponding to the 1's positions in each byte:

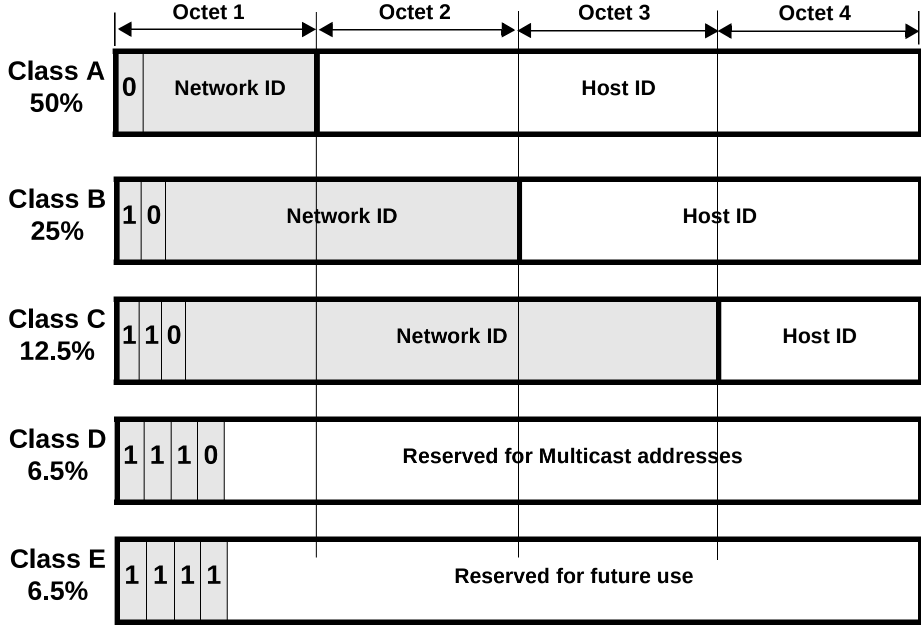
	$2^7 =$ 128	$2^6 =$ 64	$2^5 =$ 32	$2^4 =$ 16	$2^3 =$ 8	$2^2 =$ 4	$2^1 =$ 2	$2^0 =$ 1
194 =	1	1	0	0	0	0	1	0
13 =	0	0	0	0	1	1	0	1
219 =	1	1	0	1	1	0	1	1
7 =	0	0	0	0	0	1	1	1

IP Address Classes

The IP address space (2^{32} or about 4.3 billion addresses) has been divided into five groups, Classes A–E, to accommodate the need for different network sizes. Each class has a different allocation of bits between the network and host IDs. The classes are identified by a fixed pattern of leading bits.

In Class A addresses, the first (leftmost) bit is always 0. So Class A IP addresses have 7 bits to define network IDs; 7 bits can define a total of 128 (0-->127) Class A networks. The remaining 24 bits of a Class A IP address are used to define host IDs. So for each of the 126 networks, there are 2^{24} or 16,777,216 possible hosts.

The following table shows how IP addresses are the allocated among the five classes.



Address classes A, B, and C cover 87.5% of the address space. These addresses are assigned by the ISP or the Internet Assigned Number Authority (IANA) to organizations for their exclusive use. The remaining 12.5% of addresses, designated classes D and E, are reserved for special purposes.

The IANA assigns a network address to an organization and a network administrator in the organization assigns the Host IDs associated with that Network ID to nodes within the organization's network.

Table 12 shows the ranges of network and host IDs, and the total number of IP addresses (# network IDs times # host IDs), for each class.

Table 12. Network and host IDs.

Class	Network ID Range	Host ID Range	Total IP Addresses
A	7 bits 126 Networks: 1 to 126	24 bits 16.8 Million Hosts per network: 0.0.1 to 255.255.254	2.1 Billion 50%
B	14 bits 16,382 Networks: 128.0 to 191.255	16 bits 65,534 Hosts per network 0.1 to 255.254	1.1 Billion 25%
C	21 bits 2.1 Million Networks: 192.0.0 to 233.255.255	8 bits 254 Hosts per network: 1 to 254	0.5 Billion 12.5%
D&E			0.5 Billion 12.5%

You can tell the class of an IP address by the first octet. For example, 191.221.30.101 is a Class B address and 192.221.30.101 is a Class C address.

Private IP Address

Addresses on the Internet need to be unique to avoid ambiguity in message routing over the Internet. To insure uniqueness, the Internet Assigned Number Authority (IANA) controls the use of IP addresses. Organizations that maintain private networks that never communicate with the Internet can use arbitrary IP addresses as long as they are unique within the private network. To help prevent the duplication of IP addresses on the Internet, the IANA has reserved the following ranges of IP addresses for private networks:

1 Class A networks: 16.6 Million addresses: 10.0.0.0 --> 10.255.255.255

16 Class B networks: 1 Million addresses: 172.16.0.0 --> 172.31.255.255

256 Class C networks: 65,000 addresses:192.168.0.0 --> 192.168.255.255

These IP addresses can be used repeatedly in separate private networks, which are not connected to the Internet. Routing tables prohibit the propagation of these addresses over the Internet. (See RFC 1918). All other IP addresses are unique and must be assigned by the IANA or ISP.

Subnetting

Subnetting is the grouping of IP addresses associated with a network ID into two or more subnetworks. The subnets of a network ID are visible only within the organization that owns the network ID; Internet routers route messages based on the network ID and the routers within the private organization differentiate between the individual subnets.

Reasons for subnetting

Subnetting is desirable because it enables a more efficient allocation and management of IP addresses.

The three-class hierarchy of IP addresses results in an inefficient allocation of addresses in many cases because addresses are assigned and managed in blocks by network ID. For example, a company that needs 10,000 IP addresses in each of two locations might be assigned two Class B network IDs, each of which provides 65,534 IP addresses. Even though one Class B network ID would provide more than enough addresses for both locations, having a separate network ID for each location is easier to manage. If the company uses only 20,000 of these addresses, about 100,000 go unused.

In this case, subnetting would enable the company to use one Class B network ID and subdivide the addresses into two subnets, one for each location. Each subnet would have a unique “extended network ID” that would enable them to be managed as if they had unique network IDs.

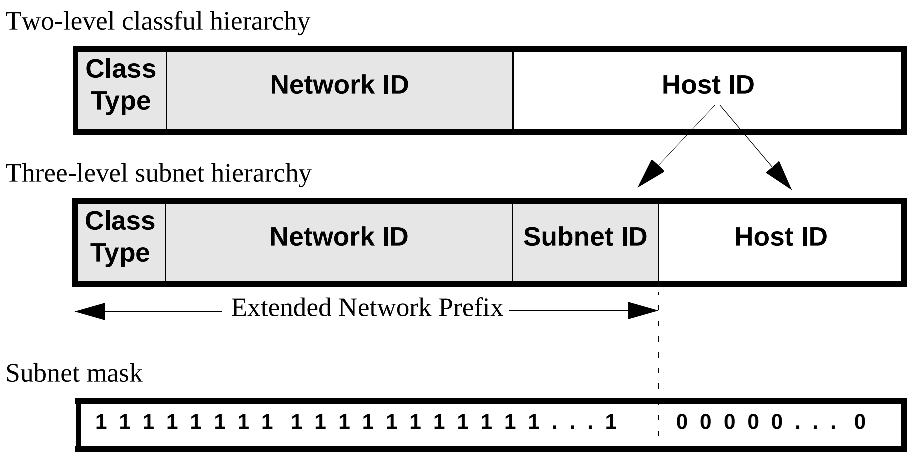
Typically, organizations need to manage IP addresses in separate groups based on several criteria in addition to location:

- different types of LANs
- different server applications
- different work projects
- security

The grouping of IP addresses provided by the three-Class structure does not allow nearly enough flexibility to meet the needs of most organizations. Subnetting allows the N IP addresses associated with a network ID to be divided into as few as 2 groups, each with N/2 addresses, or into as many as N/2 groups, each with 2 addresses, if desired.

How subnets are created

RFC 950 defines a standard procedure to divide a Class A, B, or C network ID into subnets. The subnetting adds a third level of hierarchy to the two-level hierarchy of the Class A, B, and C network ID number. An “extended network prefix” is formed by using two or more bits of the Host ID as a subnet number, and appending this subnet number to the network ID.



The extended network prefix is then treated as a normal network ID. The remaining host ID bits define the host IDs within each subnet. For example, a block of IP addresses could be subdivided into four subnets by using 2 host bits to “extend” the network ID. Now there are 4 times as many (extended) networks and 1/4 as many hosts per network.

NOTE:
In adding up the number of network and host IDs, certain addresses cannot be counted. In general, addresses with all ones or all zeros in either the network portion or the host portion of the address are not usable. These are reserved for special uses, such as broadcasting or loopback.

Subnet Masks

Routing protocols use a *subnet mask* to determine the boundary between the extended network ID and the host ID in an IP address. The subnet mask is a 32-bit binary number consisting of a string of contiguous 1’s followed by a string of contiguous 0’s. The 1’s part corresponds to the extended network prefix and the 0’s part corresponds to the host ID of the address.

Each of the three classes of addresses has a default subnet mask that specifies the end of the 1st, 2nd, and 3rd octet as the boundary between the extended network prefix and the host ID. The default subnet mask in each case means “no subnetting.”

	Default Subnet Mask
Class A	11111111.00000000.00000000.00000000 255.0.0.0
Class B	11111111.11111111.00000000.00000000 255.255.0.0
Class C	11111111.11111111.11111111.00000000 255.255.255.0

In addition to the default subnet masks, which divide the network and host IDs at the octet boundaries in the IP address, subnets can be formed by using 2 or more bits from the host octets to define the subnet ID.

Class-C subnets

Table 13 shows that Class-C IP addresses can have 5 subnetting schemes, each with a different number of subnets per network. The first and last subnet, formed by using 1 and 7 bits respectively, are unusable because they result in either the subnet ID or the host ID having all zeros or all ones.

Table 13. Subnetting schemes for Class-C IP addresses.

No. Sub-net bits	No. of Usable Subnets per NW	No. of Hosts per Subnet	No. of Usable IP Addresses	Binary Subnet ID (4th Octet)	Decimal Subnet ID	Class C Subnet Masks
1	0	126	0	10000000	128	255.255.255.128
2	2	62	124	11000000	192	255.255.255.192
3	6	30	180	11100000	224	255.255.255.224
4	14	14	196	11110000	240	225.225.225.240
5	30	6	180	11111000	248	255.255.255.248
6	62	2	124	11111100	252	255.255.255.252
7	126	0	0	11111110	254	255.255.255.254

3-bit subnets

As an example, the third row of the table shows the results of using 3 bits for the subnet ID. Three bits are “borrowed” from the host ID leaving 5 bits for the host IDs. The number of subnets that can be defined with three bits is $2^3 = 8$ (~~000~~, 001, 010, 011, 100, 101, 110, ~~111~~). Of these, only 6 are usable (all ones and all zeros are not usable). The remaining 5 bits are used for the host IDs. Of these, $2^5 - 2 = 30$ are usable. As shown in columns 2–4 (row 3), by using 3 bits for subnetting, a Class C network can be divided into 6 subnets with 30 host IDs in each subnet for a total of 6 X 30 = 180 usable IP addresses.

Subnet mask

The subnet mask is defined as follows. The subnet bits “borrowed” from the host ID are the highest-order bits in the octet of the host ID. The 5th and 6th columns of the table show the binary and decimal subnet IDs, formed by using the subnet bits as the highest-order bits in an octet. For example, in the third row of the table, the binary bit pattern is 11100000, which is decimal 224. This is the highest number that can be formed with the 3 high-order bits in the octet. The subnet mask is formed by putting this number in the 4th octet of the default subnet mask (shown in the last column of the table).

The mask, 255.255.255.224, corresponds to a bit pattern of 27 ones followed by 5 zeros. This mask would be used to check that two IP addresses are on the same or different subnets by comparing the first 27 binary digits of the two addresses. If the first 27 binary digits are the same, the two addresses are on the same subnet.

Example

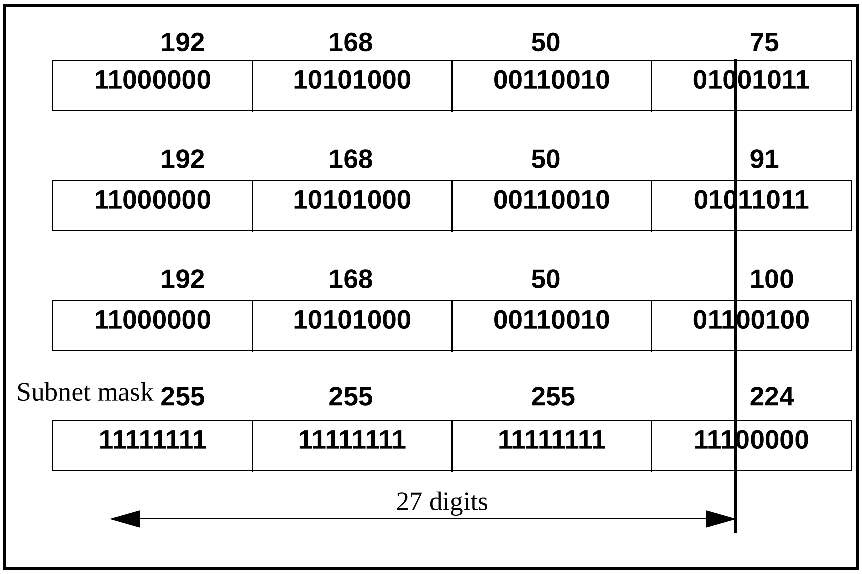
To continue the example using a 3-bit subnet ID, assume a Class C network ID of 192.168.50.xxx. This network ID can provide 254 usable IP addresses, all on the same network — from 192.168.50.1 to 192.168.50.254. If we divide this network into 3-bit subnets, we will have 6 usable subnets with 30 usable IP addresses in each subnet. Note that we have lost 74 usable IP addresses in the process because we had to discard the all-ones and all-zeros subnet IDs (62 addresses) and host IDs (12 addresses). There is always a loss of usable IP addresses with subnetting.

Table 14 shows the subnet boundaries for the six subnets formed with 3 bits. The boundaries are the numbers formed by using all combinations of 3 bits as the highest-order bits in an octet (Columns 1 and 2) and then using these numbers in the 4th octet for the host IDs.

Table 14. Subnet Boundaries

Binary Subnet Boundaries (for 3 bits)	Decimal Subnet Boundaries	Range of usable IP Addresses in the Subnet
00000000	0	not usable
00100000	32	192.168.50.**33** to 192.168.50.**62**
01000000	64	192.168.50.**65** to 192.168.50.**94**
01100000	96	192.168.50.**97** to 192.168.50.**126**
10000000	128	192.168.50.**129** to 192.168.50.**158**
10100000	160	192.168.50.**161** to 192.168.50.**190**
11000000	192	192.168.50.**193** to 192.168.50.**222**
11100000	224	not usable

For example, the IP addresses 192.168.50.75 and 192.168.50.91 are on the same subnet but 192.168.50.100 is on a different subnet. This is illustrated in the following diagram where the subnet mask, 255.255.255.244 is used to compare the first 27 binary digits or each address.



The other four possible subnetting schemes for Class C addresses, using 2, 4, 5, and 6 subnet bits, are formed in the same way. Which of the 5 subnetting schemes to use depends on the requirements for the number of subnets and the number of hosts per subnet.

Class-A and Class-B subnets

For Class A and Class B IP addresses, subnets can be formed in the same way as for Class C addresses. The only difference is that many more subnets per network can be formed. For Class B networks, subnets can be formed using from 2 to 14 bits from the 3rd and 4th octets. For Class A networks, subnets can be formed using from 2 to 22 bits from the 2nd, 3rd and 4th octets.

The Subnet Mask field on the PPP Data Module screen (used for PPP connections) and on the IP Interfaces screen (used for Ethernet connections) enables the specification of a subnet for the IP address.

Valid subnet masks

The valid subnets for each Class of address are:

Table 15. Valid subnets.

Class A (default 255.0.0.0)	Class B: (default 255.255.0.0)	Class C (default 255.255.255.0)
255.192.0.0	255.255.192.0	255.255.255.192
255.224.0.0	255.255.224.0	255.255.255.224
255.240.0.0	255.255.240.0	255.255.255.240
255.248.0.0	255.255.248.0	255.255.255.248
255.252.0.0	255.255.252.0	255.255.255.252
255.254.0.0	255.255.254.0	
255.255.0.0	255.255.255.0	
255.255.128.0	255.255.255.128	
255.255.192.0	255.255.255.192	
255.255.224.0	255.255.255.224	
255.255.240.0	255.255.255.240	
255.255.248.0	255.255.255.248	
255.255.252.0	255.255.255.252	
255.255.254.0		
255.255.255.0		
255.255.255.128		
255.255.255.192		
255.255.255.224		
255.255.255.240		
255.255.255.248		
255.255.255.252		

Notice that all 5 valid Class C subnet masks can also be valid Class B or Class A subnet masks, and all 13 valid Class B subnet masks can also be valid Class A subnet masks.

For example, `255.255.255.224` is a valid subnet mask for all three address classes. It allows 6 (2^3–2) subnetworks for Class C addresses, 2046 (2^{11}–2) subnetworks for Class B addresses and 524,286 (2^{19}–2)subnetworks for Class A addresses. Each of these subnetworks can have 30 (2^5–2) hosts.

Network IP addresses

An IP address for a network has the network ID in the network portion and 0 in the host portion. For example, 192.168.1.0 is the network address for the 192.168.1 network.

When a network is subnetted, and you want to set up a network IP route to a subnetwork, the IP address of the subnetwork is the first address in that subnetwork, which has all 0's for the host portion of the address. For example, the subnet mask formed by using 2 bits of the host portion of a Class C address is 255.255.255.192 (1100000 = 192). For the 192.168.1 network, this subnet mask creates 2 usable subnetworks whose IP addresses are 192.168.1.64 (0100000 = 64) and 192.168.1.128 (1000000 = 128), with 62 usable host addresses in each subnetwork.

Network Bits to Subnet Mask

Table 16. Network Bit and Subnet Mask comparison

Network Bits	Subnet Mask	Number of Hosts	Network Type
0	0.0.0.0	4,294,967,294	0
1	128.0.0.0	2,147,483,646	1
2	192.0.0.0	1,073,741,822	2
3	224.0.0.0	536,870,910	3
4	240.0.0.0	268,435,454	4
5	248.0.0.0	134,217,726	5
6	252.0.0.0	67,108,862	6
7	254.0.0.0	33,554,430	7
8	255.0.0.0	16,777,214	8
9	255.128.0.0	8,388,606	9
10	255.192.0.0	4,194,302	10
11	255.224.0.0	2,097,150	11
12	255.240.0.0	1,048,574	12
13	255.248.0.0	524,286	13
14	255.252.0.0	262,142	14
15	255.254.0.0	131,070	15
16	255.255.0.0	65,534	16
17	255.255.128.0	32,766	17
18	255.255.192.0	16,382	18
19	255.255.224.0	8,190	19
20	255.255.240.0	4,094	20
21	255.255.248.0	2,046	21
22	255.255.252.0	1022	22
23	255.255.254.0	510	23
24	255.255.255.0	254	24
25	255.255.255.128	126	25
26	255.255.255.192	62	26

Continued on next page

Table 16. Network Bit and Subnet Mask comparison *(Continued)*

27	255.255.255.224	30	27
28	255.255.255.240	14	28
29	255.255.255.248	6	29
30	255.255.255.252	2	30

Glossary

A

AAR
See Automatic Alternate Routing (AAR).

Abbreviated Dialing (AD)
A feature that allows callers to place calls by dialing just one or two digits.

AD
See Abbreviated Dialing (AD)

ADU
See Asynchronous data unit (ADU).

adjunct
A processor that does one or more tasks for another processor and that is optional in the configuration of the other processor. Intuity AUDIX and CMS are considered adjuncts to the Avaya MultiVantage solution.

Administration Without Hardware (AWOH)
A feature that allows administration of ports without associated terminals or other hardware.

ANSI
American National Standards Institute**.** A United States professional/technical association supporting a variety of standards.

APLT
Advanced Private-Line Termination.

ARP
Address Resolution Protocol.

ARS
See Automatic Route Selection (ARS).

ASCII
American Standard Code for Information Interchange. The standard code for representing characters in digital form. Each character is represented by an 8-bit code (including parity bit).

asynchronous data transmission
A method of transmitting data in which each character is preceded by a start bit and followed by a stop bit, thus permitting data characters to be transmitted at irregular intervals. This type transmission is advantageous when transmission is not regular (characters typed at a keyboard). Also called asynchronous transmission.

Asynchronous data unit (ADU)
A device that allows direct connection between RS-232C equipment and a digital switch.

Asynchronous Transfer Mode (ATM)
A connection-oriented, digital service optimized for fiber-optic lines at speeds up to 622.08 mbps. ATM networks set up a virtual circuit (virtual connection) between the transmitter and the receiver before sending any data. Data is then sent in a continuous stream of fixed-length, 58-byte cells, each of which contains a 48-byte payload and a 5-byte header. The header contains the virtual circuit number that identifies the pre-negotiated path through the network.

ATM
See Asynchronous Transfer Mode (ATM)

Audio Information Exchange (AUDIX)
A fully integrated voice-mail system. Can be used with a variety of communications systems to provide call-history data, such as subscriber identification and reason for redirection.

Automatic Alternate Routing (AAR)
A feature that routes calls to other than the first-choice route when facilities are unavailable.

Automatic Route Selection (ARS)
A system for automatically routing telephone calls by the least costly route.

auxiliary trunk
A trunk used to connect auxiliary equipment, such as radio-paging equipment, to a communications system.

AWOH
See Administration Without Hardware (AWOH).

B

B8ZS
See Bipolar 8 Zero Substitution (B8ZS).

Basic Rate Interface (BRI)
A standard ISDN frame format that specifies the protocol used between two or more communications systems. As used in North America, BRI provides 23 64-kbps B-channels (voice or data) and one 64-kbps D-channel (signaling). The D-channel is the 24th channel of the interface and contains multiplexed signaling information for the other 23 channels.

bandwidth
The amount of data that a given channel can transmit in a given period of time, measured in bits per second (not bytes per second) on digital networks or in Hertz (cycles per second) on analog networks. For analog transmission, the band width can be measured as the difference, expressed in hertz, between the highest and lowest frequencies transmitted.

baud
A unit of transmission rate equal to the number of signal events per second. See also bit rate.

BCC
See bearer capability class (BCC)

bearer capability class (BCC)

Code that identifies the type of a call (for example, voice and different types of data). Determination of BCC is based on the caller's characteristics for non-ISDN endpoints and on the Bearer Capability and Low-Layer Compatibility Information Elements of an ISDN endpoint. Current BCCs are 0 (voice-grade data and voice), 1 (DMI mode 1, 56 kbps data transmission), 2 (DMI mode 2, synchronous/asynchronous data transmission up to 19.2 kbps) 3 (DMI mode 3, 64 kbps circuit/packet data transmission), 4 (DMI mode 0, 64 kbps synchronous data), 5 (temporary signaling connection, and 6 (wideband call, 128–1984 kbps synchronous data).

BER

Bit error rate.

Bipolar 8 Zero Substitution (B8ZS)

B8ZS line coding substitutes a mix of 1s and 0s for every group of eight consecutive 0s in a stream of AMI-encoded data (see line coding). The encoded string contains consecutive ones with the same polarity. These intentional, bipolar violations of the AMI coding scheme let the receiving end identify, decode, and restore the long zero strings in the original message. B8ZS line coding does not corrupt digital data, so it is commonly used with T-1 lines.

bit (binary digit)

One unit of information in binary notation, having two possible values: 0 or 1.

bit rate

The speed at which bits are transmitted, usually expressed in bits per second. Also called data rate.

bps

Bits per second.

BOS

Bit-oriented signaling.

BRI

See Basic Rate Interface (BRI)

bridge

A device that connects two or more packet-switched networks and directs packets sent from one to the other. See router.

bus bridge

A connection between the TDM bus and the packet bus built into the C-LAN circuit pack for use with DEFINITY ECS csi. Bus bridge connectivity is not used with any other DEFINITY switch model.

BX.25

A version of the CCITT X.25 protocol for data communications. BX.25 adds a fourth level to the standard X.25 interface. This uppermost level combines levels 4, 5, and 6 of the ISO reference model.

byte

A sequence of (usually eight) bits processed together.

C

Call Detail Recording (CDR)
A feature that uses software and hardware to record call data. (Same as station message detail recording — SMDR).

Call Detail Recording utility (CDRU)
Software that collects, stores, optionally filters, and outputs call-detail records.

Call Management System (CMS)
An application, running on an adjunct processor, that collects information from an ACD unit. CMS enables customers to monitor and manage telemarketing centers by generating reports on the status of agents, splits, trunks, trunk groups, vectors, and VDNs, and enables customers to partially administer the ACD feature for a communications system.

call redirection
See restricted facilities.

CALLed Party Number IE
The ISDN information element containing the digits sent to the called party.

capture rate
The number of frames contained in a one-second video sample, used as a measure of video quality.

CA-TSC
Call-Associated Temporary Signaling Connection.

CCIS
Common-Channel Interoffice Signaling

CCITT
Comitte Consultatif International Telephonique et Telegraphique. Now called *International Telecommunications Union* (ITU).

CCMS
Control-Channel Message Set.

CDR
See Call Detail Recording (CDR). (Same as SMDR and CMDR).

CDRU
Call Detail Record Unit.

CDRP
Call Detail Record Poller.

CEPT1
European Conference of Postal and Telecommunications Rate 1.

channel
A communication path linking two points for transmitting voice and data. Also:

1. A circuit-switched call.
2. In wideband, all of the time slots (contiguous or noncontiguous) necessary to support a call. Example: an H0-channel uses six 64-kbps time slots.
3. A DS0 on a T1 or E1 facility not specifically associated with a logical circuit-switched call; analogous to a single trunk.

circuit

1. An arrangement of electrical elements through which electric current flows.
2. A channel or transmission path between two or more points.

circuit pack

A card on which electrical circuits are printed, and IC chips and electrical components are installed. A circuit pack is installed in a switch carrier.

Class of Restriction (COR)

A feature that allows up to 64 classes of call-origination and call-termination restrictions for voice terminals, voice-terminal groups, data modules, and trunk groups.

Class of Service (COS)

A feature that uses a number to specify if voice-terminal users can activate the Automatic Callback, Call Forwarding All Calls, Data Privacy, or Priority Calling features.

circuit-switched network

A network that sets up and maintains a connection for the exclusive use of two or more communicating parties for the duration of their call. The familiar, voice telephone network is circuit-switched. See packet switching.

clear-channel facility

A digital circuit that requires no in-channel framing or control bits. The whole bandwidth is thus available for data transmission.

client

An application that runs on one processor while drawing on data or other resources that are on a server located elsewhere. **MMCX client:** a workstation capable of making MMCX calls. Such a workstation is a client of one or more MMCX servers. See dial-plan table.

CMDR

Centralized Message Detail Recording. (Same as CDR and SMDR).

CMS

Call Management System.

COR

See Class of Restriction (COR).

COS

See Class of Service (COS).

CP

Circuit pack.

CSN

See circuit-switched network.

CSU

Channel service unit.

cyclic redundancy checking (CRC)

A method for detecting read, transmit, and write errors in data. At the transmission end, the system treats a block of data as a single binary number, divides it by some specified binary number, and appends the remainder (called the CRC character) to the data. At the receiving end, the system recalculates the remainder and compares the result to the CRC character. If the two agree, there are no errors.

D

data channel
A communications path between two points used to transmit digital signals.

data-communications equipment (DCE)
The equipment (usually a modem, data module, or packet assembler/disassembler) on the network side of a communications link that makes the binary serial data from the source or transmitter compatible with the communications channel.

datagram
In packet switching, a packet that carries information sufficient for routing from the originating data terminal equipment (DTE) without the necessity of establishing a connection between the DTEs and the network. Connectionless, unreliable.

data link
The configuration of physical facilities enabling end terminals to communicate directly with each other.

data path
The end-to-end connection used for a data communications link. A data path is the combination of all elements of an interprocessor communication in a DCS.

data port
A point of access to a computer that uses trunks or lines for transmitting or receiving data.

data service unit (DSU)
A device that transmits digital data on transmission facilities.

data terminal equipment (DTE)
Equipment consisting of the endpoints in a connection over a data circuit. In a connection between a data terminal and host, the terminal, the host, and their associated modems or data modules make up the DTE.

DCE
Data-communications equipment.

D-channel backup
Type of backup used with Non-Facility Associated Signaling (NFAS). A primary D-channel provides signaling for an NFAS D-channel group (two or more PRI facilities). A second D-channel, on a separate PRI facility of the NFAS D-channel group, is designated as backup for the D-channel. Failure of the primary D-channel causes automatic transfer of call-control signaling to the backup D-channel. The backup becomes the primary D-channel. When the failed channel returns to service, it becomes the backup D-channel.

DCIU
Data communications interface unit.

DCP
Digital Communications Protocol.

DCS
Distributed Communications System.

dial-plan table
A data structure that defines how a switch or server (such as MMCX) interprets dialed digits and routes calls. The dial-plan table performs two tasks. First, it identifies a dial plan rule that applies to the kind of input it has received. Then it applies the rule and translates the dialed input into a corresponding extension or public-network telephone number. (MMCX interserver calls are handled differently; see interserver routing table.)

DID
Direct Inward Dialing.

digital communications protocol (DCP)
A proprietary protocol used to transmit both digitized voice and digitized data over the same communications link. A DCP link is made up of two 64-kbps information (I-) channels and one 8-kbps signaling (S-) channel. The DCP protocol supports 2 information-bearing channels, and thus two telephones/data modules.

digital signal level 0 (DS0)
A single 64-kbps voice channel. A DS0 is a single 64-kbps channel in a T1 or E1 facility and consists of eight bits in a T1 or E1 frame every 125 microseconds.

digital signal level 1 (DS1)
A single 1.544-Mbps (United States) or 2.048-Mbps (outside the United States) digital signal carried on a T1 transmission facility. A DS1 converter complex consists of a pair, one at each end, of DS1 converter circuit packs and the associated T1/E1 facilities.

digital terminal data module (DTDM)
An integrated or adjunct data module that shares with a digital telephone the same physical port for connection to a communications system. The function of a DTDM is similar to that of a PDM and MPDM in that it converts RS-232C signals to DCP signals.

distributed application
A computer application that runs on one or more clients and uses shared resources, such as databases. These resources reside on a common server. Distributed design lets multiple users run programs using common, centrally maintained files.

domain
An addressable location on a network, such as a group of computers, single computer, or subdirectory. See Domain Name Server (DNS).

Domain Name Server (DNS)
An Internet computer that maintains a database of domain names.

Provides a mapping of alphanumeric names to IP addresses; for example, xxx.xxx.xxx.xxx --> www.avaya.com.

DNS
See Domain Name Server (DNS).

DSU
Data service unit.

DTDM
Digital-terminal data module.

DTE
Data-terminal equipment.

E

E-1
A digital transmission link with a capacity of 2.048 Mbps (2,048,000 bits per second). The European equivalent of the T-1. It can support 30 multiplexed 64-Kbps voice and data channels plus separate 64-Kbps channels for signalling and framing (synchronization). Also spelled **E1**.

EIA
See Electronics Industries Association (EIA).

EIA-232
A physical interface specified by the EIA. EIA-232 transmits and receives asynchronous data at speeds of up to 19.2 kbps over cable distances of up to 50 feet. EIA-232 replaces RS-232 protocol in some applications.

electronic tandem network (ETN)
A tandem tie-trunk network that has automatic call-routing capabilities based on the number dialed and the most preferred route available. Each switch in the network is assigned a unique private network office code (RNX), and each voice terminal is assigned a unique extension.

Electronics Industries Association (EIA)
A trade association of the electronics industry that establishes electrical and functional standards.

Ethernet
A local area network (LAN) that works over short distances on twisted-pairs or coaxial cables at speeds up to 10 mbps or 100 mbps. One of the two LAN protocols MMCX supports. See ATM, LAN emulation (LANE).

Ethernet Source Address
A 48-bit physical address of the NIC; also called the MAC address.

ETN
Electronic tandem network

F

facility
A telecommunications transmission pathway and associated equipment.

facility-associated signaling (FAS)
Signaling for which a D-channel carries signaling only for those channels on the same physical interface.

FAS
Facility-associated signaling

framing
The data-formatting conventions that allow a receiver to synchronize with the transmitting end of a circuit. For example, T-1 frames contain an 8-bit sample from each of the 24 channels on the interface (192 bits total) plus a framing bit (for a total of 193 bits). Each framing bit marks the end of a timed sample the input at the transmission end.

FRL
Facilities Restriction Level.

FX
Foreign exchange.

G

Gateway
(1) protocol converter (2) a node between network segments.

H

H.320
The most common standard for videoconferencing over ISDN BRI circuits. H.320-compatible systems can communicate with each other even when they rely on dissimilar hardware and software.

H.323
A specification that sets standards for multimedia communications between LANs and telephony networks, such as ISDN.

HDB3
See High Density Bipolar 3-Bit Substitution (HDB3).

High Density Bipolar 3-Bit Substitution (HDB3)
HDB3 line coding is similar to Bipolar 8 Zero Substitution (B8ZS) in some ways. It replaces every 4 consecutive zero in a stream of AMI-encoded data (see line coding) with either of two sequences. If there has been an even number of 1s since the last substitution, it substitutes the pattern **1 0 0 *BipolarViolation***, where ***BipolarViolation*** is a 3-volt pulse (a **1**) of the same polarity as the preceding 3-volt pulse. If there has been an odd number of 1s since the last substitution, HDB3 coding substitutes the pattern **0 0 0 *BipolarViolation*** for the 4-zero string. This system does not corrupt binary data, and is commonly used with E-1 lines.

host
A server.

host name
See server name.

I

IANA
Internet Assigned Number Authority.

INADS
Initialization and Administration System

Information element (IE)
The data fields in ISDN messages.

in-service state
The condition or state of an MMCX server that is ready to handle calls.

International Standards Organization (ISO)
A body that defines and/or adopts protocols widely used in the computer and telecommunications industries.

International Telecommunications Union (ITU)
Formerly known as International Telegraph and Telephone Consultative Committee (CCITT), ITU is an international organization that sets universal standards for data communications, including ISDN. ITU members are from telecommunications companies and organizations around the world.

Internet
The decentralized network of networks that grew from ARPAnet and supported by TCP/IP.

interserver routing table
A database that keeps track of the interserver routing tables on a multiserver MMCX network. When a server sets up a Point-to-Point Protocol connection with a remote server, the local server looks up the dialed-digit sequences in the interserver routing table. (Calls that involve only one MMCX server are routed using the dial-plan table.)

I/O base address
The place in memory where a given computer peripheral places and picks up messages that travel over a computer's system bus.

IP
Internet Protocol.

IP (Internet Protocol) address
A 32-bit number that uniquely identifies endpoints on the Internet, commonly specified in the form ***n1.n2.n3.n4*** where each n_n is a decimal number between **0** and **255**. Part of the IP address represents the address of a local network's gateway to the Internet and part represents the host-machine address within that local network. The available bits are apportioned to the network address or local address using a system of classes. The Class A addresses used by the largest organizations on the Internet reserve the first 8 bits for the network portion of the address and remaining 24 for the host machine. Class B addresses, the most common class, assign 16 bits to the network and 16 to the host machine. The Class C addresses used by small networks reserve the first 24 bits for the network and the remaining 8 bits for the host.

IP user
An H.323 endpoint on an MMCX network. Such users do not have log ins, so the MMCX server identifies them using an extension number and an IP address.

ISDN
Integrated Services Digital Network, a digital, voice and data service. You get ISDN in either of two forms. Primary Rate Interface (PRI) service has a capacity of 1.544 mbps, divided into 23 or 29 B-channels (23 on a North American T-1 connection, 29 on a European E-1) and 1 D-channel, each with a capacity of 64 kbps. Basic Rate Interface (BRI) service has a capacity of 144 kbps, divided into 2 B-channels at 64 kbps each and 1 D-channel at 16 kbps.

ISDN trunk
A trunk administered for use with ISDN-PRI. Also called ISDN facility.

L

LAN
See local area network.

LAN emulation (LANE)

A technique that lets ATM networks communicate with Ethernet LAN cards. ATM service is not widely available at the desktop and ATM interface cards are expensive, so ATM networks usually emulate LANs when communicating with user terminals. Typically, a **LAN-emulation configuration server (LECS)** keeps track of the relationships between ATM-network addresses and IP addresses on the LAN. Each ATM card is then treated as a client of a **LAN-emulation server**, which connects the ATM cards to the LAN cards on the user terminals.

LAP-D

See link-access procedure on the D-channel (LAPD).

LECS (LAN-emulation configuration server)

See LAN emulation (LANE).

LES (LAN-emulation server)

See LAN emulation (LANE).

line coding

Line coding is the data format that lets either end of a communications channel correctly interpret messages from the other. Line coding systems specify the voltage levels and patterns that represent binary digits (1s and 0s), based on the requirements of the transmission network. The AT&T network has two: it demands that the net voltage on the line equal 0 volts DC and it demands a minimum Open System Interconnect (OSI) Model. The T-carrier system meets the first requirement by using a bipolar line-coding scheme called Alternate Mark Inversion (AMI). It meets the second with one of several supplementary coding schemes, including ZCS, Bipolar 8 Zero Substitution (B8ZS), and High Density Bipolar 3-Bit Substitution (HDB3).

line compensation

An allowance for pulse distortions in the cable that connects the MMCX server to the first channel service unit (CSU) on the PRI span. It is proportional to the length of the cable. T1 circuit packs adjust the outgoing signal so that it arrives at the far end without distortion.

link

A transmitter-receiver channel that connects two systems.

link-access procedure on the D-channel (LAPD)

A link-layer protocol on the ISDN-BRI and ISDN-PRI data-link layer (level 2). LAPD provides data transfer between two devices, and error and flow control on multiple logical links. LAPD is used for signaling and low-speed packet data (X.25 and mode 3) on the signaling (D-) channel and for mode-3 data communications on a bearer (B-) channel.

local area network

A short-range data communication network providing high-speed carrier service with low error rates. Generally, a LAN is limited in range to a maximum of 6.2 miles. Ethernet and Token-Ring are common LAN architectures. See wide area network, Ethernet.

looparound testing

Checking an interface by sending a signal through the output, across a medium (the loop), and back through the input. Internal looparound tests run against the internal circuitry of the card. External looparound tests check the connectors at the edge of the card using a fiber or wire loop. PRI, Ethernet, and ATM cards are tested this way.

M

MAC
Media Access Control; a sublayer of the OSI Data Link layer, provides an interface with the network adapter.

main-satellite-tributary
A private network configuration that can either stand alone or access an ETN. A main switch provides interconnection, via tie trunks, with one or more subtending switches, called satellites; all attendant positions for the main/satellite configuration; and access to and from the public network. To a user outside the complex, a main/satellite configuration appears as one switch, with one listed directory number (LDN). A tributary switch is connected to the main switch via tie trunks, but has its own attendant positions and LDN.

Management Information Base (MIB)
A virtual database used by the Simple Network Management Protocol (SNMP).

MAPD
Multiapplication platform for DEFINITY.

MA-UUI
Message-Associated User-to-User Signaling.

M-Bus
Memory bus.

MDR
Message detail record.

MIB
See Management Information Base (MIB).

MMCH
Multimedia call handling.

MSA
Message Server Adjunct.

multicasting
A transmission method that promotes efficient bandwidth utilization on a multimedia LAN when several parties are transmitting and receiving simultaneously. Normally, each party sends a separate video stream to each of the other parties and receives a separate video stream from each in return (this is called unicasting). Multicasting substitutes a single broadcast transmission (addressed to all parties) for the separate transmissions addressed to each. Each party then continues to receive multiple data streams while sending only one. For best results, multicast messages should be restricted to one or more subnet.

N

National Television Standards Committee (NTSC) standard
The standard format and transmission method for television signals in North America, Central America, and Japan. The NTSC is a division of the Electronic Industries Association (EIA). MMCX video transmissions conform with NTSC requirements.

national number
The full dialed number minus the country code. The national number is composed of office code plus subscriber code. In North America, this means area code plus seven digits. For example, 303-538-1234 is a national number.

NCA/TSC
Noncall-associated/temporary-signaling connection.

NCOSS
Network Control Operations Support Center.

NCSO
National Customer Support Organization.

NEC
National Engineering Center.

NEMA
National Electrical Manufacturer's Association.

NETCON
Network-control circuit pack.

Network
A collection of computer-like devices ("nodes") that are connected by, and can communicate across, a common transmission medium.

Network Adapter
The interface between a node and the network; has a unique physical address.

Network Interface Card (NIC)
A circuit board that can be fitted to a personal computer (PC) to allow the PC to communicate with other machines on a network. MMCX works with Ethernet cards and with asynchronous transfer mode (ATM) cards running Ethernet emulation. Also called a network adapter.

NFAS
See Nonfacility-associated signaling (NFAS).

NIC
See Network Interface Card (NIC).

Node
1. A point on a network that provides an interface to a communications device.

2. A switch or adjunct in an Avaya DCS network.

Nonfacility-associated signaling (NFAS)
A method that allows multiple T1 and/or E1 facilities to share a single D-channel to form an ISDN-PRI. If D-channel backup is not used, one facility is configured with a D-channel, and the other facilities that share the D-channel are configured without D-channels. If D-channel backup is used, two facilities are configured to have D-channels (one D-channel on each facility), and the other facilities that share the D-channels are configured without D-channels.

NTSC
See National Television Standards Committee (NTSC) standard.

O

Open System Interconnect (OSI) Model

An International Standards Organization (ISO) interoperability specification. OSI defines standard services that compliant communications networks and equipment must provide, rather than specific implementations. It divides network operations into 7 steps, called layers, and arranges them hierarchically, in a protocol stack. The rules (protocols) in each layer of the stack specify a service that other parts of a communications system can always get, as long as they make their requests in a standard way. This approach leaves designers free to implement the internal details of the service in whatever way seems best to them. The OSI layers are **1 Physical** (transmission medium), **2 Datalink** (link-level signaling and error control), **3 Network** (computer-to-computer signaling, routing, etc.), **4 Transport** (delivery, end-to-end error control, and flow control), **5 Session** (dialog management), **6 Presentation** (data-format compatibility), and **7 Application** (file-transfer services, virtual terminals, etc.).

OSI

See Open System Interconnect (OSI) Model.

out-of-service state (OOS)

The condition or state of an MMCX server that is operating but not ready to accept or place calls.

P

PACCON

Packet controller.

packet

A group of bits (including a message element, which is the data, and a control information element (IE), which is the header) used in packet switching and transmitted as a discrete unit. In each packet, the message element and control IE are arranged in a specified format.

packet bus

A wide-bandwidth bus that transmits packets.

packet switching

A data-transmission technique whereby user information is segmented and routed in discrete data envelopes called packets, each with its own appended control information, for routing, sequencing, and error checking. The packets can travel to their destinations by varying routes. For data transmissions, a packet switched network can make more efficient use of available bandwidth than a circuit-switched network, because it does not dedicate a channel for the duration of a call. Instead, packets are queued and sent on a standby basis, as channel capacity becomes available. The Internet is a good example of a packet-switching network.

PBX

Private Branch Exchange: a customer-owned telephone switch that connects a company's internal telephone network with the local telephone service provider's central office.

PDM

See processor data module (PDM).

PGATE

Packet gateway.

Point-to-Point Protocol
A TCP/IP implementation tailored for use over telephone lines. It supports router-to-router and host-to-network connections over both synchronous and asynchronous circuits. PPP replaces SLIP, the older Serial Line Interface Protocol.

Port
(1) Interface between an application and the TCP/IP network. A port is a predefined internal address (port number) that serves as a pathway from the application to the Transport layer (or from T to A). (2) circuit-pack port.

PPP
See Point-to-Point Protocol.

PRI
Primary Rate Interface. See ISDN.

Primary Rate Interface
See ISDN.

private network
A network used exclusively for the telecommunications needs of a particular customer.

private network office code (RNX)
The first three digits of a 7-digit private network number.

processor data module (PDM)
A device that provides an RS-232C DCE interface for connecting to data terminals, applications processors (APs), and host computers, and provides a DCP interface for connection to a communications system.

protocol
A set of conventions or rules governing the format and timing of message exchanges to control data movement and correction of errors.

PSDN
Packet-switch public data network.

PSTN
See Public Switched Telephone Network (PSTN).

PSN
Packet-switched network.

Public Switched Telephone Network (PSTN)
The worldwide voice telephone system.

Q

QPPCN
Quality Protection Plan Change Notice.

QSIG
A set of open standards for Enterprise Networking. QSIG is a protocol defining message exchanges (signalling) at the “Q” reference point between two PBXs.

R

RBS
Robbed-bit signaling.

RFC
Request for comment.

restricted facilities
PRI spans that use ZCS line coding, the opposite of unrestricted facilities.

RNX
Route-number index (private network office code)

router
An interface between different networks. Routers support network management, including load balancing, route optimization, prioritizing of calls, and troubleshooting. They are thus more capable than bridges.

routing plan
Routing plans direct outgoing voice-interworking and interserver calls to the first available PRI trunk group in a list. This arrangement lets you allow for trunk groups that are busy, out of service, or out of bandwidth. Each routing plan is identified by a routing-plan number in the range **1-32**.

RPN
Routing-plan number.

RTP
Real-time protocol.

S

Socket
An address formed by concatenating the IP address and the port number.

server
1. Any system that maintains and administers files that are used by independent, client applications. 2. the MMCX server, the computer that sets up, maintains, and administers MMCX network communications. See dial-plan table.

server name
The name that identifies the MMCX server on your local area network. You enter the server name in the boot-time administration menu. See server number.

server number
The ISDN international number that uniquely identifies your MMCX server on the PBX and/or the public telephone network. See server name, interserver routing table.

signaling

The control information that a network uses to set up and maintain connections. On-hook and off-hook are, for instance, the familiar voice-telephone signals that tell the central office that you have picked up the telephone handset or hung up at the end of a call.

In-channel signalling reserves part of the available data -communication bandwidth for control information (see restricted facilities). Out-of-channel signalling schemes use a separate channel for signals, so that data transmissions can use all of the bandwidth available to them (see clear-channel facility).

Simple Network Management Protocol (SNMP)

An International Standards Organization (ISO) protocol that sets standards for communications between network components and network management stations. SNMP handles network resources using the information contained in a management information base (MIB), a virtual database resident on the various parts of the network. SNMP supports security, configuration, performance, fault management, and accounting management. SNMP is part of the ISO Open System Interconnect (OSI) Model specification.

SMDR

Station Message Detail Recording. (Same as call detail recording — CDR).

SNMP

See Simple Network Management Protocol (SNMP).

SPE

Switch Processing Element

subnet

A network within a larger network. Subdividing a network into subnets improves the efficiency of routing and makes the best use of the limited number of addresses available with any given addressing scheme. Subnetworks are defined by using part of the host-machine portion of the IP (Internet Protocol) address as an additional layer of network information. Routers can then distinguish between a networking part that is significant for internet/intranet routing and a host-machine part, which is significant only within the local subnet.

subnet mask

A bit pattern that lets a network administrator define subnets using the host-machine portion of the IP (Internet Protocol) address. The subnet mask has binary ones in positions corresponding to the network and subnet parts of the address and zeros in the remaining, host-address positions. During IP-address resolution, zero fields in the mask hide corresponding host-address fields in the address, causing the router to ignore them. The router resolves only the networking part of the address, leaving the host part for the local subnet to resolve. This increases speed and makes multicasting more efficient. Subnet masks are usually written in the decimal notation used for IP addresses.

T

T-1

A 4-wire (2 twisted pair), digital communications link with a capacity of 1.544 Mbps (1,544,000 bits per second). A T-1 provides 24 separate 64-Kbps channels. These can support up to 23 concurrent voice and data channels plus a separate channel for signalling and framing (synchronization). T-1 is the standard for data communications in North America and Japan. Also spelled **T1**. See E-1.

T-carrier
A hierarchy of digital voice- and data-transmission systems used in North America and based on multiples of the capacity of the T-1 line.

tandem switch
A switch within an electronic tandem network (ETN) that provides the logic to determine the best route for a network call, possibly modifies the digits outpulsed, and allows or denies certain calls to certain users.

tandem-through
The switched connection of an incoming trunk to an outgoing trunk without human intervention.

tandem tie-trunk network (TTTN)
A private network that interconnects several customer switching systems.

TCP
Transmission Control Protocol — Transport layer; Connection oriented, reliable.

TCP/IP
Transmission Control Protocol/Internet Protocol. A standard that lets different computer hardware and different operating systems (such as PCs, Apple computers, UNIX workstations, and mainframes) communicate with each other over a network. TCP/IP is the most complete, most widely accepted network protocol currently available.

TDM
See time-division multiplexing.

tie trunk
A telecommunications channel that directly connects two private switching systems.

time-division multiplexing
A way of interleaving digitized voice, video, and/or data so that several calls can be sent concurrently over the same spread of frequencies. TDM systems distribute the available bandwidth across a series of time slots and divide each input stream into a set of samples. They then assign successive time slots to the first sample in each input stream. They repeat the process until all time slots have been assigned or all inputs sent. At the far end, the receiver demultiplexes the transmission and reconstructs the original sequence of samples. Time-division multiplexing is typically used on circuit-switched networks.

time slot
64 kbps of digital information structured as eight bits every 125 microseconds. In the switch, a time slot refers to either a DS0 on a T1 or E1 facility or a 64-kbps unit on the TDM bus or fiber connection between port networks.

trunk
A communication line linking two switches (public-network, PBX, or MMCX-server).

trunk group
A collection of trunks that all terminate at the same public switch, PBX, or MMCX server.

U

UDP
1. User Datagram Protocol — Transport layer; Connectionless, unreliable, fast.

2. Uniform Dial Plan.

UNP
Uniform numbering plan

unrestricted data
Data that has to travel over an unrestricted facilities, because it can be corrupted by Zero Code Suppression (ZCS) line coding. Digital data is unrestricted data.

unrestricted facilities
PRI spans that use non-ZCS line coding, such as B8ZS or HDB3, end-to-end. See Zero Code Suppression (ZCS).

V

VCI
The ATM virtual channel identifier. ATM is a connection-oriented service that sets up a virtual circuit between the transmitter and the receiver. The two endpoints negotiate an agreed route through the network before sending any data. VCI and VPI together constitute the virtual circuit number that identifies this route.

virtual channel identifier
See VCI.

virtual circuit (virtual connection)
The pre-arranged route through the ATM network that all cells in an ATM transmission follow.

virtual circuit number
The route identifier contained in an ATM cell header. A VCI and a VPI together constitute the virtual circuit number.

virtual path identifier
See VPI.

VPI
The ATM virtual path identifier. ATM is a connection-oriented service that sets up a virtual circuit between the transmitter and the receiver. The two endpoints negotiate an agreed route through the network before sending any data. VCI and VPI together constitute the virtual circuit number that identifies this route. The VPI is analogous in some ways to an ISDN trunk group.

W

WAN
See wide area network.

well-known address
The default network address for a given type of network component, as specified by a communications protocol or standards body. For example, the ATM Forum defines well-known addresses for LAN-emulation servers and LAN-emulation configuration servers.

well-known port
port numbers that are assigned to specific applications by IANA. For example, 21 for FTP, 23 for Telnet, 110 for pop3.

wide area network

A data network that connects local area networks (LANs) using common-carrier telephone lines, bridges, and routers.

wideband

A circuit-switched call at a data rate greater than 64 kbps. A circuit-switched call on a single T1 or E1 facility with a bandwidth between 128 and 1536 (T1) or 1984 (E1) kbps in multiples of 64 kbps. H0, H11, H12, and N x DS0 calls are wideband.

Z

ZCS

See Zero Code Suppression (ZCS).

Zero Code Suppression (ZCS)

ZCS line coding substitutes a 1 for the second least-significant bit of every all-zero byte in AMI-encoded data (see line coding). ZCS encoding has no effect on voice communications, but it corrupts digital data (see restricted facilities). In MMCX communications, ZCS corrupts the ISDN D-channel.

Index

Numerics

A

B

C

D

E

F

G

H

I

J

L

M

N

O

P

Q

R

S