Administration for Network Connectivity for Avaya Communication Manager

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About this document

Purpose

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

Changes from the previous version

This version of Administration for Network Connectivity, Issue 12, has been changed to focus more closely on procedural information related to IP telephony implementation. The following changes were made:

- Modification of chapter 1, "Networking Overview." Most of the information previously in this chapter is duplicated in other documents and has been condensed or removed. A large section providing descriptions of port network connectivity (PNC) configurations, including detailed diagrams, has been added to this chapter and removed from the Hardware Description document.
- Update of chapters 3 and 4, "Administering Converged Networks" and "Network Quality Administration." These chapters have been updated for release 4.0 of Communication Manager.
- Removal of chapters 5 and 6: "Administering Dedicated Networks" and "Feature Interactions and Considerations." These chapters contain mostly descriptive information about circuit-switched Definity networks and can be accessed in the previous version of this book, Issue 11, February 2006.
- Removal of Appendixes A and B: "Using IP Routes" and Internet Control Message Protocol (ICMP) ECHO messages." The material in these appendixes is unchanged for the release 4.0 of Communication Manger and can be accessed in the previous version of this book, Issue 11, February 2006.

The descriptive material that was removed from this version of Administration for Network Connectivity is available in the previous version: Issue 11, February 2006. Some of the removed material is duplicated by similar material in:

- *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207
- *Avaya Application Solutions: IP Telephony Deployment Guide*, 555-245-600

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: Control Networks for S8700-Series and S8500 Media Servers provides information on how to set up control networks.
- Chapter 3: Administering converged networks provides procedures for initial administration of server-to-gateway connections, including a sample network configuration procedure with administration screens, IP trunks using H.323 IP connections, DCS AUDIX and CMS adjunct administration, installing and administering Avaya IP telephones, and administering IP-to-IP connections.
- Chapter 4: Network quality administration provides instructions for administering Quality of Service on telephony and network equipment.

Chapter 1: Networking overview

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

About “network” terminology

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

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

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

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

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

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

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

About digital telephone calls

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

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

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

About network regions

A network region is a group of IP endpoints that share common characteristics and resources. Every IP endpoint on an Avaya Communication Manager system belongs to a network region.

By default, all IP endpoints are in network region 1. If left that way, all IP endpoints would all share the same characteristics defined by network region 1 and use the same resources. But in many cases, this is not sufficient to allow for certain differences that may be based upon location or network characteristic, and therefore multiple network regions should be configured.

The most common of these cases are:

- One group of endpoints requires a different CODEC (COder-DECoder) set than another group.

  This could be based on requirements related to bandwidth or encryption.
- Calls between separate groups of endpoints require a different codec set than calls within a single group of endpoints, again based on requirements related to bandwidth or encryption.
- Specific C-LAN or MedPro or other resources must be accessible to only a specific group of endpoints.

- One group of endpoints requires a different UDP port range or QoS parameters than another group.
- One group of endpoints reports to a different VoIP Monitoring Manager server than another group.

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

Establishing inter-switch trunk connections

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

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

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

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

These and other common trunk types are described in the *Administrator Guide for Avaya Communication Manager*, 03-300509.

Interconnecting port networks

Note:

See Port network configurations with S8500 and S8700-series Media Servers on page 23 for detailed examples of IP-connected (IP-PNC) and fiber-connected (fiber-PNC) port networks.

Avaya systems with more than three fiber-connected port networks (fiber-PNC, formerly called "Multi-Connect") must use a center stage switch (CSS) or an ATM configuration to interconnect the port networks.

Networking branch offices

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

For more information, see: http://www.multitech.com/PARTNERS/Alliances/Avaya/.

Control networks

Control networks are the networks over which media servers, such as the S8700-series or S8500 Media Servers, exchange signaling data with the port networks through the IPSI circuit packs.

With Communication Manager 3.0 and later, Avaya extends “Control Network on Customer LAN” functionality to simplify network configuration by allowing both IP-PNC and fiber-PNC port networks in a single configuration. With this combined port network functionality, enterprises can attach IP-connected, ATM-connected, or center-stage-connected port networks to their S8700-series or S8500 media servers.

To support combined port networks, Avaya has enhanced the flexibility of control networks for port network attachment. In addition to private control networks A and B, Avaya allows the “Customer LAN” Ethernet interface to be used as a third, public control network, control network C.

Note:

See Chapter 2: Control Networks for S8700-Series and S8500 Media Servers on page 103 for more information about control networks.

Enabling spanning tree protocol (STP)

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

However, STP is slow to converge after a network failure, and slow to allow a new port into the network (~50 sec by default).

A modified version of STP, Rapid Spanning Tree converges faster than the earlier STP, and enables new ports much faster (sub-second) than the older protocol. **Rapid Spanning Tree** works with all Avaya equipment, and is *recommended* by Avaya.

Inter-Gateway Alternate Routing (IGAR)

For single-server systems that use the IP-WAN to connect bearer traffic between port networks or media gateways, Inter-Gateway Alternate Routing (IGAR) provides a means of alternately using the PSTN when the IP-WAN is incapable of carrying the bearer connection. IGAR may request that bearer connections be provided by the PSTN under the following conditions:

- The number of calls allocated or bandwidth allocated via Call Admission Control-Bandwidth Limits (CAC-BL) has been reached.
- VoIP RTP resource exhaustion in a MG/PN is encountered.
- A codec set is not specified between a network region pair.
- Forced redirection between a pair of network regions is configured.

IGAR takes advantage of existing public and private-network facilities provisioned in a network region. Most trunks in use today can be used for IGAR. Examples of the better trunk facilities for use by IGAR would be:

- Public or Private ISDN PRI/BRI
- R2MFC

IGAR provides enhanced Quality of Service (QoS) to large distributed single-server configurations.

Dial Plan Transparency

Dial Plan Transparency (DPT) preserves the dial plan when a media gateway registers with an LSP or when a port network registers with an ESS due to the loss of contact with the primary controller. DPT establishes a trunk call and reroutes the call over the PSTN to connect endpoints that can no longer connect over the corporate IP network.

Network quality management

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

- *Delay:* Significant end-to-end delay may result in echo and talker overlap.
- *Packet Loss:* Under peak network loads and periods of congestion, voice data packets may be dropped.
- *Jitter (Delay Variability):* Jitter results when data packets arrive at their destination at irregular intervals as a result of variable transmission delay over the network.

Note:

For more information about these QOS factors and network quality management, see:

- Chapter 4: Network quality administration on page 201
- *Avaya Application Solutions: IP Telephony Deployment Guide*, 555-245-600

About VoIP-transmission hardware

The following circuit packs are essential in an Avaya telecommunications network.

For more information about these and other Avaya hardware devices, see *Hardware Description and Reference for Avaya Communication Manager*, 555-245-207.

For information about the administration tasks for this equipment, see Setting up VoIP hardware on page 117.

- TN799DP control LAN (C-LAN) interface

  The TN799DP control LAN (C-LAN) interface provides TCP/IP connectivity over Ethernet between media servers and gateways or Point to Point Protocol (PPP) between media servers and adjuncts.

- TN2312BP IP Server Interface (IPSI)

  The IPSI provides for the transport of control messages between media servers and port networks.

- TN2302AP IP Media Processor and TN2602AP IP Media Resource 320

  The TN2302AP and TN2602AP provide high-capacity VoIP audio access to the switch for local stations and outside trunks.

- TN8400AP Media Server circuit pack

  The TN8400 Media Server circuit pack is the hardware platform for an S8400 Media Server, which is a Linux-based server that occupies a single slot in a standard TN carrier.

- TN8412AP S8400 server IP Interface (SIPI)

  The SIPI is used in an S8400-based system to provide transport of control messages between the S8400 Media Server and the media server's port network (PN) using direct connections.

- H.248 media gateways

  The H.248 media gateways include the G700, G250, G350, G860, and IG550.

  The H.248 media gateways provide:

  - Extension of Communication Manager telephony features to branch offices when controlled by a remote media server.

- Standalone telephony systems when controlled by an embedded S8300 Media Server.
- Local Survivable Processor (LSP) backup for a remote media server.

- MM760 VoIP Media Module

  The MM760 VoIP Media Module is a clone of the G700 motherboard VoIP engine. The MM760 provides an additional 64 VoIP channels in the G700.

Processor Ethernet (PE)

Much like a C-LAN board, Processor Ethernet provides connectivity to IP endpoints, gateways, and adjuncts. The PE interface is a logical connection in the Communication Manager software that uses a port on the NIC in the server (that is, the s-called “native NIC”). No additional hardware is needed to implement PE. Processor Ethernet uses the PROCR IP-interface type.

During the configuration of a server, the PE is assigned to a Computer Ethernet (CE). The PE and the CE share the same IP address, but are very different in nature. The CE interface is a native computer interface while the PE interface is the logical appearance of the CE interface within Communication Manager software. The interface that is assigned to the PE can be a control network or a corporate LAN. The interface that is selected determines which physical port the PE uses on the server. For more information on how to configure the server, see the *Administrator Guide for Avaya Communication Manager,* 03-300509.

The PE interface is enabled automatically on a Local Survivable Processor (LSP) or an Enterprise Survivable Server (ESS). On an LSP, the H.248 and the H.323 fields default to a yes on the **ip-interface procr** screen to allow the registration of H.248 gateways and H.323 endpoints using the PE interface. While the PE interface on a simplex ESS provides support for adjunct connectivity, it does not support H.248 gateway and H.323 endpoint registration. Therefore the H.248 and H.323 fields on the ESS’ **ip-interface procr** screen default to a no.

Note:

The PE interface can be enabled but not administered with no adverse effects on the system.

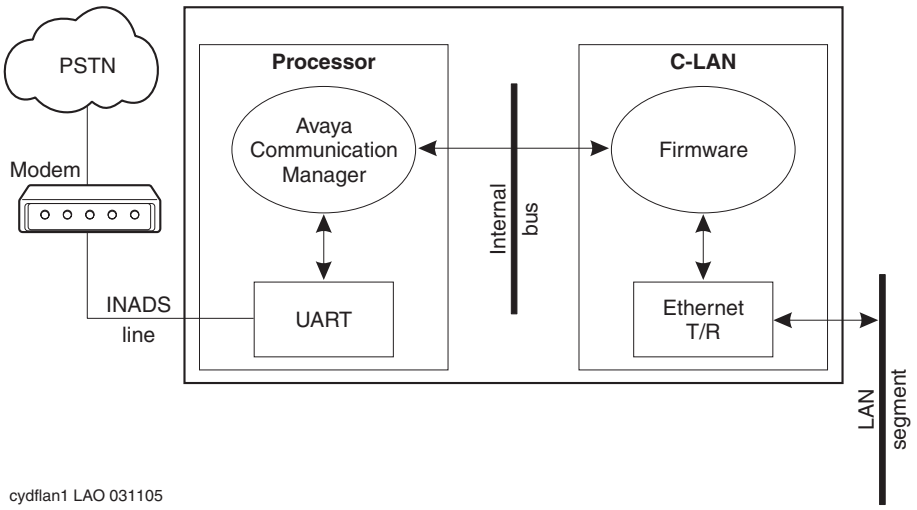
CAUTION:

Both the ESS and the LSP require the use of the PE interface to register to the main call server. Do not disable the PE interface on an ESS server or an LSP.

Providing LAN security

Some customers are concerned that a user could access the switch using the INADS line, gain access to C-LAN, and then access to the customer's LAN. The Avaya architecture prevents access to the customer's LAN as depicted in Figure 1: Security-related system architecture on page 20, which shows a high-level switch schematic with a TN799 (C-LAN) circuit pack.

Figure 1: Security-related system architecture



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

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

  The user can only enter SAT commands that request C-LAN information or to configure C-LAN connections.

- The C-LAN application TFTP is currently disabled and cannot be enabled by Avaya Communication Manager.

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

Connection Preservation

The Connection Preserving Migration (CPM) feature preserves existing bearer (voice) connections while an H.248 media gateway migrates from one Communication Manager server to another because of network or server failure. However, users on connection-preserved calls cannot use such features as Hold, Conference, or Transfer, etc. In addition to preserving the audio voice paths, CPM extends the time period for recovery operations and functions during Avaya's complementary recovery strategies:

- H.248 and H.323 Link Recovery
- Auto fallback to primary
- Local Survivable Processor (LSP)
- Enterprise Survivable Server (ESS)
- Standard Local Survivability (SLS) on the G250 Media Gateway only

H.248 and H.323 Link Recovery

H.248 Link Recovery is an automated way in which the media gateway reacquires the H.248 link when it is lost from either a primary call controller or an LSP. The H.248 link between a media server running Avaya Communication Manager and a media gateway, and the H.323 link between a media gateway and an H.323-compliant IP endpoint, provide the signaling protocol for:

- Call setup
- Call control (user actions such as Hold, Conference, or Transfer) while the call is in progress
- Call tear-down

If the link goes down, Link Recovery preserves any existing calls and attempts to re-establish the original link. If the gateway/endpoint cannot reconnect to the original server/gateway, then Link Recovery automatically attempts to connect with alternate TN799DP (C-LAN) circuit packs within the original server's configuration or to a Local Survivable Processor (LSP).

Auto fallback to primary

The intent of the auto fallback to primary controller feature is to return a fragmented network, in which a number of H.248 Media Gateways are being serviced by one or more LSPs (Local Survivable Processors), to the primary media server in an automatic fashion. This feature is targeted towards all H.248 media gateways. By migrating the media gateways back to the primary automatically, the distributed telephony switch network can be made whole sooner without human intervention, which is required today.

Local Survivable Processor (LSP)

Either an S8300 or S8500 Media Server can act as survivable call-processing servers for remote or branch customer locations. As an LSP, the S8300 Media Server carries a complete set of Communication Manager features, and its license file allows it to function as a survivable call processor. If the link between the remote G700/G350 media gateway(s) and the primary controller is broken, those telephones and media gateways that are designated to receive backup service from the LSP will register with the LSP. The LSP will provide control to those registered devices in a license error mode (see *Hardware Description and Reference for Avaya Communication Manager,* 555-245-207).

Note:

The LSP, in contrast to the Standard Local Survivability (SLS) feature on the G250 Media Gateway, is also known as ELS, or Enhanced Local Survivability.

Enterprise Survivable Server (ESS)

The Enterprise Survivable Server (ESS) feature provides survivability to port networks by allowing backup servers to be placed in various locations in the customer's network. The backup servers supply service to port networks in the case where the Avaya S8500 Media Server, or S8700-series Media Server pair fails, or connectivity to the main Communication Manager server(s) is lost. Servers for ESS can be either S8500 or S8700-series media servers, and offer full Avaya Communication Manager functionality when in survivable mode, provided sufficient connectivity exists to other Avaya components (for example, endpoints, gateways, and messaging servers).

Standard Local Survivability (SLS)

Standard Local Survivability (SLS) consists of a module built into the G250 Media Gateway to provide partial backup media gateway controller functionality, in the event that the connection with the primary controller is lost. This feature allows a G250, with no S8300 installed locally, to provide a degree of Communication Manager functionality when no link is available to an external controller. It is configured on a system-wide basis, or, alternatively, it can be configured on an individual G250 using the CLI.

Port network configurations with S8500 and S8700-series Media Servers

The S8500 and S8700-series Media Servers can control call processing of port networks in a large variety of ways. Control networks can be established using Ethernet connections only or a combination of Ethernet connections and fiber connections (direct-connect, CSS, or ATM). Voice, fax, TTY, and modem transmission can occur over the LAN/WAN connections, fiber connections, or both. Reliability with the S8700-series Media Server can include single control and bearer networks (standard reliability), duplicated control networks (high reliability), duplicated control and bearer networks (critical reliability), or a combination of reliabilities.

Each of the following configurations show how the various options can be used. Configurations with both IP-PNC and fiber-PNC PNs on page 79 describes the possibilities and considerations when fiber-PNC options are combined with IP-PNC options.

Fiber-PNC and IP-PNC

Fiber port network connectivity (fiber-PNC) uses fiber connections and/or DS1-C connections between port networks (PNs) for the following:

- Voice/data bearer transmission
- Control signaling from the server to PNs that do not have a control TN2312BP IPSI circuit pack
- Sharing of Touch-tone Receiver (TTR) and media processor resources. If these resources are not available in one fiber-PNC PN, these resources on another fiber-PNC PN can be used across the fiber links.

Fiber-PNC includes Direct Connect, Center Stage Switch (CSS), and ATM configurations. One or more PNs in the CSS or ATM configurations have an IPSI connection to the server for control signaling. Only one PN in a Direct Connect configuration has an IPSI connection.

IP port network connectivity (IP-PNC) uses LAN/WAN connections exclusively between port networks for bearer transmission and control signaling from the server. Each PN must have either one or two control ISPI circuit packs for control signaling.

An S8500 or S8700-series Media Server can support both types of port network connectivity simultaneously within a single Communication Manager configuration.

Reliability

Reliability is the ability of a Communication Manager configuration to maintain service when components such as Ethernet switches, circuit packs, or gateways within the configuration fail. The available reliability levels and their precise definitions depend on whether the port networks use IP-PNC or fiber-PNC and whether the server is an S8500 or S8700-series Media Server.

S8500 Media Server

An S8500-series Media Server has several reliability options, and the implementation of reliability levels differs somewhat between fiber-PNC and IP-PNC.

Fiber-PNC

- Standard reliability

  For fiber-PNC, an S8500 Media Server supports only Direct Connect with up to 3 fiber-PNC PNs. This configuration supports a single control IPSI in one of the fiber-PNCs. IPSIs in other PNs serve only as tone clocks and do not carry control signaling. There can be only a single fiber connection between each of the fiber PNs.

IP-PNC

- Standard reliability

  For IP-PNC, an S8500 Media Server supports a single IPSI for control in every IP-PNC PN. TN2302BP or TN2602AP circuit packs are used for the bearer network. However, TN2602AP circuit packs are implemented in load-balancing mode only.

- Duplicated bearer reliability

  For IP-PNC, an S8500 Media Server does not support duplicated control. However, any or all IP-PNC PNs may have duplicated TN2602AP circuit packs to duplicate the bearer connections. Control signaling to a PN with duplicated TN2602AP circuit packs always occurs over a direct IPSI connection to the server. Duplicated bearer using TN2602AP circuit packs is implemented for individual PNs and does not require uniform implementation for all PNs within the configuration.

S8700-series Media Server

An S8700-series Media Server has multiple levels of reliability, and the implementation of reliability levels differs somewhat between fiber-PNC and IP-PNC.

Fiber-PNC

All port networks that use fiber-PNC within a single Communication Manager configuration must have the same level of reliability and may be one of the following:

- Standard duplex reliability — The standard S8700-series Media Server configuration includes duplicated servers. A single IPSI circuit pack for control resides in one or more PNs. A single fiber interface connects all fiber-PNC PNs.
- High reliability — The high reliability S8700-series Media Server configuration includes the standard duplicated servers. In addition, duplicated IPSIs for control reside in each PN designated to have control IPSIs. A single fiber interface connects all fiber-PNC PNs.
- Critical Reliability — The standard reliability S8700-series Media Server configuration includes the following:
  - Standard duplicated servers
  - Duplicated IPSIs for control reside in each PN designated to have control IPSIs.
  - Duplicated fiber interfaces connect all fiber-PNC PNs.

IP-PNC

Reliability for PNs that use IP-PNC within a single Communication Manager configuration is implemented for individual PNs and does not require uniform implementation for other IP-PNC PNs within the configuration. In addition, duplicated bearer and duplicated control can be implemented independently of each other. Duplicated control is not required in order for a PN to have duplicated bearer reliability.

An IP-PNC PN can have one of the following reliability levels:

- Standard duplicated servers

  A single IPSI provides control signaling to and from the server and there are no duplicated TN2602AP circuit packs to duplicate the bear connections, only single or load balancing TN2302BP and/or TN2602AP circuit packs.

- Duplicated control

  In addition to the standard duplicated servers, duplicated IPSIs for control reside in each PN. The PN contains only single or load balancing TN2302BP and/or TN2602AP circuit packs.

- Single control and duplicated bearer

  In addition to the standard duplicated servers, duplicated TN2602AP circuit packs reside in each PN. to provide duplicated bearer.

  Note:

  Duplicated IPSI control is recommended, but not required, for duplicated bearer for IP-PNC PNs.

- Duplicated control and bearer

  In addition to the standard duplicated servers, duplicated IPSIs for control reside in each PN and duplicated TN2602AP circuit packs reside in each PN to provide duplicated bearer.

S8500 IP-PNC (single control network)

In this configuration, the S8500 Media Server uses IP connections to both control call processing on the port networks (PNs) and to send voice between PNs over an IP network. An existing VoIP-ready IP infrastructure can be used. This solution saves customers the cost of building a separate telephony network. In this type of configuration, all PNs are connected to the server and to each other over the customer's network. Up to 64 PNs can be configured in an IP-PNC network. Depending on the type of Ethernet switches used to connect PNs, the number of PNs, and the PN locations in the LAN and WAN, the network may require multiple Ethernet switches to support the PNs.

Only the G650 media gateway is available for new installations. However, because different migrations from older systems are supported, the following media gateways can be used in an IP-PNC network:

- G650 media gateway

  A G650 PN can consist of one to five G650 gateways in a stack connected by a TDM/LAN bus cable. One gateway, serving as control gateway in position A at the bottom of the stack, contains the following:

  - TN2312BP IPSI circuit pack

- G600 media gateway

  A PN can consist of one to four G600 gateways in a stack connected by a TDM/LAN bus cable. One gateway, serving as control gateway in position A at the bottom of the stack, contains the following:

  - TN2312AP/BP IPSI circuit pack

  Note:

  The TN2314 Processor and TN744E Call Classifier and Tone Detector circuit packs, needed for the S8100 model, are *not* used and must be removed if the G600 is being migrated from an S8100 Media Server. All gateways are port gateways, though the bottom gateway (serving as control cabinet A) contains the IPSI circuit pack.

- CMC1 media gateway

  A PN can consist of one to three CMC1 gateways in a stack connected by a TDM/LAN bus cable. One gateway, serving as control gateway in position A at the bottom of the stack, contains the following:

  - TN2312AP/BP IPSI circuit pack

  Note:

  The TN795 processor board, needed for the CSI model, is *not* used and must be removed if the CMC1 is being migrated from a DEFINITY server. The CMC1 or CMC1 stack may not be used with additional PNs.

IP/TDM conversion resource - Each PN must contain at least one TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 circuit pack. The TN2302AP or TN2602AP circuit pack provides IP-TDM voice processing of endpoint connections between PNs. These circuit packs may be inserted in any gateway in the PN. Each PN may optionally house a TN799DP C-LAN circuit pack for control of the G150 Media Gateway, the H.248 media gateways (G700, G350, G250), IP endpoints, adjunct systems such as messaging, and firmware downloads.

Ethernet connections. - In the IP-PNC configuration, the S8500 Media Server connects to the media gateways through a single Ethernet switch. Each PN also has a connection to the S8500 Media Server through a local Ethernet switch. As a result, remote PNs in an IP-PNC configuration over a WAN, which normally requires routers to complete the connection, may require their own Ethernet switches, in addition to the Ethernet switch that supports the S8500 Media Server. IP connections to the S8500 Media Server may be administered as dedicated private LAN connections or connections over the customer LAN.

Duplicated TN2602AP circuit packs in IP-PNC PNs

For an S8500 Media Server, any individual IP-PNC PN can contain load-balancing or duplicated TN2602AP circuit packs. However, TN2602AP circuit packs do *not* need to be implemented uniformly within the system. Thus, some PNs may have a single TN2602AP circuit pack, some PNs may have load-balancing TN2602AP circuit packs, and some PNs may have duplicated TN2602AP circuit packs. Thus, an S8500 Media Server can have duplicated bearer connections, even though it does not support duplicated control.

Figure 2: S8500 IP-PNC

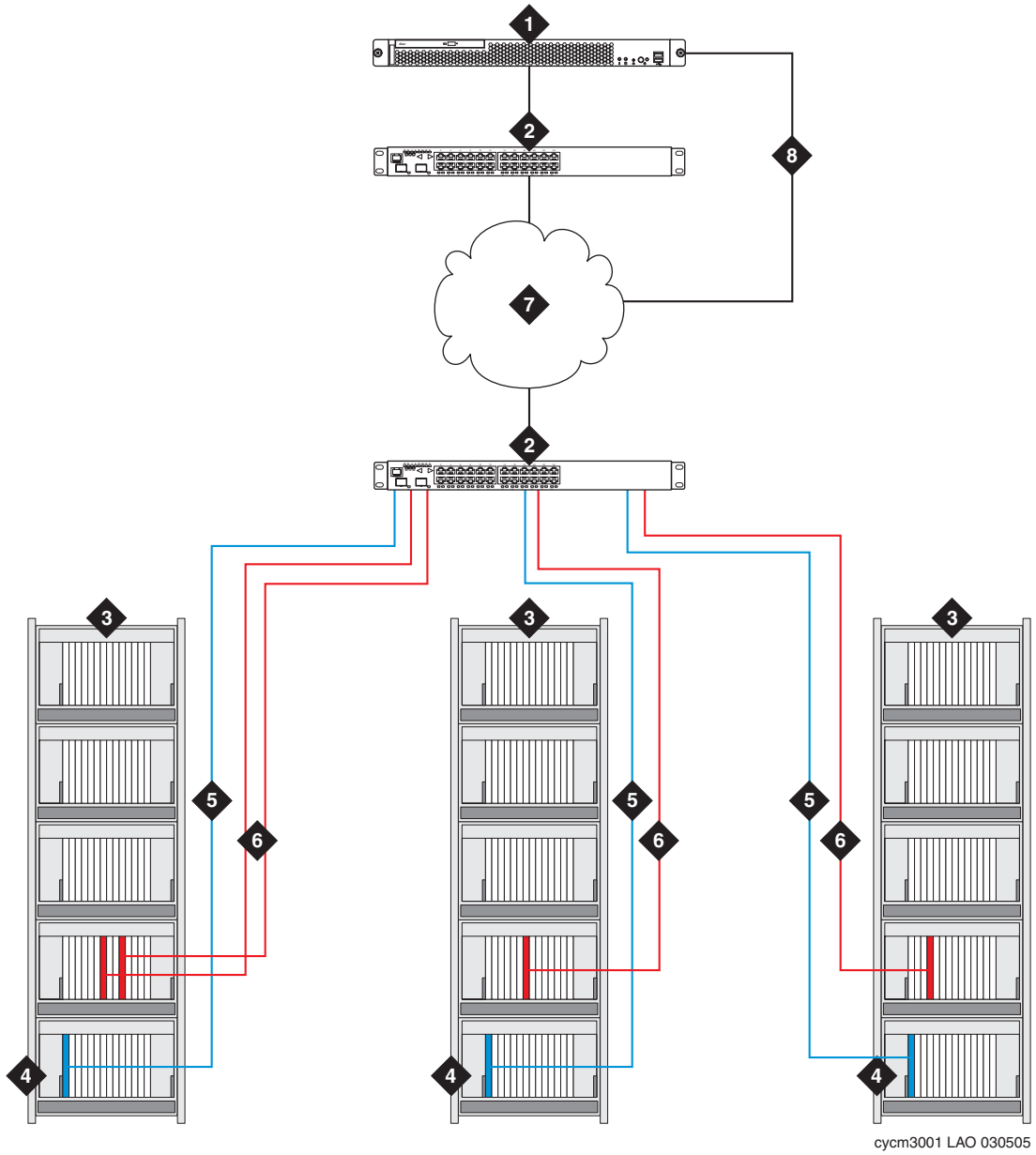


Figure notes: S8500 IP-PNC

1. S8500C or S8500B Media Server
2. Ethernet Switch. For local LAN connections, the same Ethernet switch may connect both the media servers and the media gateways. For remote LAN/WAN connections the remote gateway(s) must have an Ethernet switches at the remote location.
3. PNs (G650 Media Gateway or stack [shown in figure]). May also be a G600 or CMC1 Media Gateway or stack from an S8100 or DEFINITY Server CSI migration, an MCC1 Media Gateway from a DEFINITY Server SI or R migration, or an SCC1 Media Gateway.

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Figure notes: S8500 IP-PNC (continued)

4. PN control gateway in the A position in the gateway stack which contains:

- A TN2312AP/BP IPSI circuit pack for IP connection to server.

  NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.

5. IPSI-to-server control network connection via Ethernet switch

6. LAN connections of TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints

NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, PNs, and adjunct systems. These circuit packs may be inserted into a port gateway (shown in figure) or the PN control gateway.

7. Customer LAN/WAN

8. LAN connections of media servers for remote administration

2 of 2

S8500 direct-connect (single control network)

In this configuration, one PN is connected to the server over an Ethernet connection. Fiber links connect up to two additional PNs to each other. This configuration also requires either a dual-NIC card in the S8500 Media Server or an interim Ethernet switch so that the S8500 Media Server can have an Ethernet port to the customer LAN and a dedicated Ethernet connection to the media gateways.

IPSI-connected PN

Only the G650 media gateway is available for new installations. However, because different migrations from older systems are supported, the PN connected to the S8500 Media Server can consist of one of three gateways:

- G650 media gateway

  A G650 PN can consist of one to five G650 gateways in a stack connected by a TDM/LAN bus cable. One gateway, serving as control gateway in position A at the bottom of the stack, contains the following:

  - TN2312BP IPSI circuit pack

- SCC1 media gateway

  An SCC1 PN can consist of one to four SCC1 gateways in a stack connected by a TDM/LAN bus cable. One gateway, serving as control gateway in position A at the bottom of the stack, contains the following:

  - TN2312AP/BP IPSI circuit pack

Note:

The TN2404 and TN2401 processor circuit packs, needed for the SI model, are *not* used and must be removed if the SCC1 is being migrated from a DEFINITY server.

- MCC1 media gateway

  An MCC1 PN has from one to five carriers in an MCC1 gateway connected by a TDM/LAN bus cables. One carrier, serving as control carrier in position A in the middle of the stack, contains the following:

  - TN2312AP/BP IPSI circuit pack

  Note:

  The control carrier for a DEFINTY Server SI or R is not used and must be removed and replaced with an expansion control carrier if the MCC1 is being migrated. The processor circuit packs, needed for the SI or R models, are not used and must be removed. Other PNs can also be MCC1 Gateways.

PNs not IPSI-connected

In a S8500 direct-connect configuration, additional PNs (up to two only) may be connected using fiber optic cable. The additional PNs connect to the IPSI-connected PN using fiber optic cable between external interface (EI) TN570B (version 7 or later) circuit packs. The cables are connected to the circuit packs using short-range or long-range multi-mode transceivers, or single-mode transceivers, depending on the distance between PNs.

The TN570B circuit packs reside in the control carrier (MCC1) or control gateway (G650 or SCC1) of each PN. The control carrier or gateway in each additional PN also must contain a TN2182C Tone Clock circuit pack (SCC1 or MCC1) or a maintenance-only TN2312BP IPSI circuit pack (G650).

Note:

Straight fiber connections between TN570B circuit packs may be up to 200 feet (61 meters) (see TN570B Expansion Interface PN connections up to 200 ft. on page 75). If the distance between PNs is greater, Light guide interface units (LIUs) must also be used to connect the fiber cables or the connection must use TN1654 DS1 converters. Lengths of fiber, including connections through LIUs or DS1 converters, are:

- 4900 feet (1493.5 meters) (see TN570B Expansion Interface PN connections up to 4900/25000 ft. and 22 miles. on page 76)
- 25,000 feet (7620 meters) in multimode (see TN570B Expansion Interface PN connections up to 4900/25000 ft. and 22 miles. on page 76)
- 21.7 miles (34.9 kilometers) in single mode (see TN570B Expansion Interface PN connections up to 4900/25000 ft. and 22 miles. on page 76)
- 200 miles (322 kilometers) (see TN1654 DS1 Converter/TN570B Expansion Interface PN connections up to 200 miles. on page 78)

Note:

You *cannot* connect additional PNs that contain CMC1 or G600 Media Gateways.

TN2602AP circuit packs for duplicated bearer

For an S8500 Media Server, any individual fiber-PNC PN can contain load-balancing or duplicated TN2602AP circuit packs. However, TN2602AP circuit packs do *not* need to be implemented uniformly within the system. Thus, some PNs may have no TN2602AP circuit pack, some PNs may have load-balancing TN2602AP circuit packs, and some PNs may have duplicated TN2602AP circuit packs. Thus, an S8500 Media Server can have duplicated bearer connections, even though it does not support duplicated control.

Rules for TN570B circuit pack placement with SCC1/MCC1 Media Gateways

Fiber-PNC MCC1 and SCC1 Media Gateways have rules on the placement of TN570B External Interface circuit packs in direct connect configurations. See Rules for TN570B circuit pack placement with SCC1/MCC1 Media Gateways on page 50. However, for MCC1/SCC1 Media Gateways configured with an S8500 Media Server, only the rules that apply to single control networks apply.

Figure 3: S8500 direct-connect

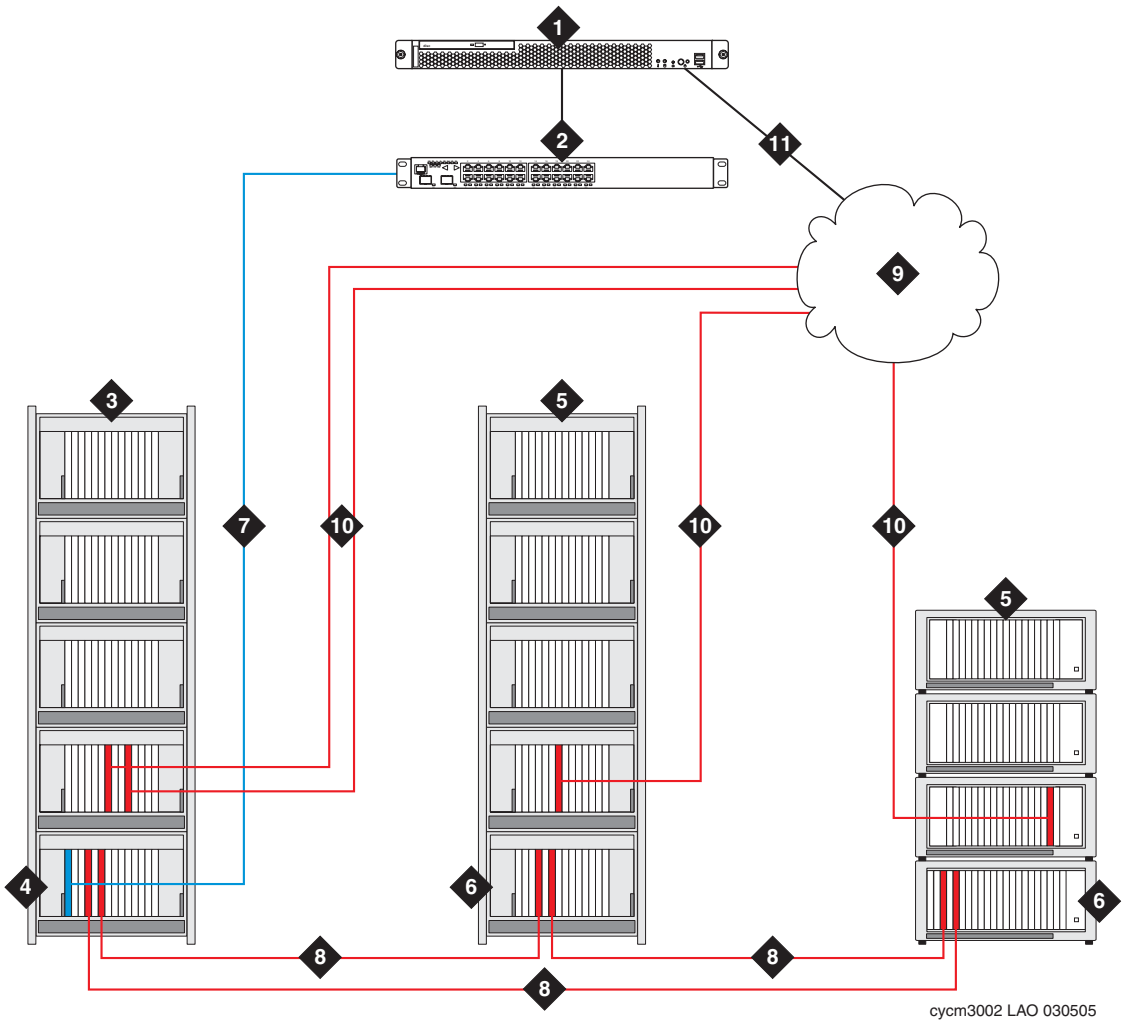

Figure notes: S8500 direct-connect

1. S8500C or S8500B Media Server
2. LAN connections of media server for remote administration
3. IPSI-connected PN (G650 Media Gateway or G650 stack [shown in figure], MCC1 Media Gateway or SCC1 Media Gateway or SCC1 stack).

   NOTE: G600 or CMC1 Media Gateways can be used in IP-PNC configurations only.
4. Media gateway (G650) or expansion port network (EPN) control gateway (SCC1) or carrier (MCC1), in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.

     NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.
   - Two TN570B EI circuit packs for bearer and control network connections to the other two PNs (if any).
5. PN (G650 Media Gateway or G650 stack [shown in figure], MCC1 Media Gateway, or SCC1 Media Gateway or SCC1 stack).

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Figure notes: S8500 direct-connect (continued)

6. PN control gateway or carrier, which contains two TN570B EI circuit packs for bearer and control network connections to the other two PNs.

   NOTE: One TN2182C Tone Clock circuit pack must also be present per PN if the PN(s) consist of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312BP IPSI circuit pack must be present per PN if the PN(s) consist of G650 Media Gateways.

   The control gateway or carrier is always in the A position in the MCC1 or gateway stack.

7. IPSI-to-server control network connection. Requires dual NIC card on the media server.

8. TN 570B/570B fiber connections between PNs

9. Customer LAN

10. LAN connections, if any, of optional TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing

    NOTE: At least one TN799DP C-LAN may optionally be present for the system for control of IP endpoints, adjunct systems such as messaging, and firmware downloads.

    NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, PNs, and adjunct systems. These circuit packs are optional for PNs in a direct-connect network and may be inserted into a port carrier (shown in figure) or the PN control carrier. However, the C-LAN circuit pack is required for downloads of firmware updates.

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S8700-series IP-PNC (single control network)

In this configuration, the S8700-series Media Servers connect to one PN or multiple PNs over an Ethernet connection using either an interim Ethernet switch and a dedicated LAN connection or the customer's LAN. Each PN is connected to the Ethernet switch or LAN with a CAT5 cable to a TN2312AP/BP IP Server Interface (IPSI) card.

This solution saves customers the cost of building a separate telephony network. In this type of configuration, all PNs are connected to the customer's network and call control from the S8700-series Media Server is also sent over the customer's network. Up to 64 PNs can be configured in an IP-PNC network.

Only the G650 media gateway is available for new installations. However, because different migrations from older systems are supported, the following media gateways can be used in an IP-PNC network:

- G650 media gateway

  A G650 PN can consist of one to five G650 gateways in a stack connected by a TDM/LAN bus cable. One gateway, serving as control gateway in position A at the bottom of the stack, contains the following:

  - TN2312BP IPSI circuit pack

- G600 media gateway

  A PN can consist of one to four G600 gateways in a stack connected by a TDM/LAN bus cable. One gateway, serving as control gateway in position A at the bottom of the stack, contains the following:

  - TN2312AP/BP IPSI circuit pack

  Note:

  The TN2314 Processor and TN744E Call Classifier and Tone Detector circuit packs, needed for the S8100 model, are *not* used and must be removed if the G600 is being migrated from an S8100 Media Server. All gateways are port gateways, though the bottom gateway (serving as control cabinet A) contains the IPSI circuit pack.

- CMC1 media gateway

  A PN can consist of one to three CMC1 gateways in a stack connected by a TDM/LAN bus cable. One gateway, serving as control gateway in position A at the bottom of the stack, contains the following:

  - TN2312AP/BP IPSI circuit pack

  Note:

  The TN795 processor board, needed for the CSI model, is *not* used and must be removed if the CMC1 is being migrated from a DEFINITY server. The CMC1 or CMC1 stack may not be used with additional PNs.

IP/TDM conversion resource - Each PN must contain at least one TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 circuit pack. The TN2302AP or TN2602AP circuit pack provides IP-TDM voice processing of endpoint connections between PNs. At least one TN799DP C-LAN circuit pack may optionally be present for control of the G150 Media Gateway, the H.248 media gateways (G700, G350, G250), IP endpoints, adjunct systems such as messaging, and firmware downloads. These circuit packs may be inserted in any gateway in the PN.

Ethernet connections. - In the IP-PNC configuration, the S8700-series Media Server connects to the media gateways through a single Ethernet switch. Each PN also has a connection to the network or the S8700-series Media Server through a local Ethernet switch. As a result, remote PNs in an IP-PNC configuration over a WAN, which normally requires routers to complete the connection, may require their own Ethernet switches in addition to the Ethernet switch that supports the S8700-series Media Server. IP connections to the S8700-series Media Server may be administered as dedicated private LAN connections or connections over the customer LAN.

Figure 4: S8700-series IP-PNC single control network

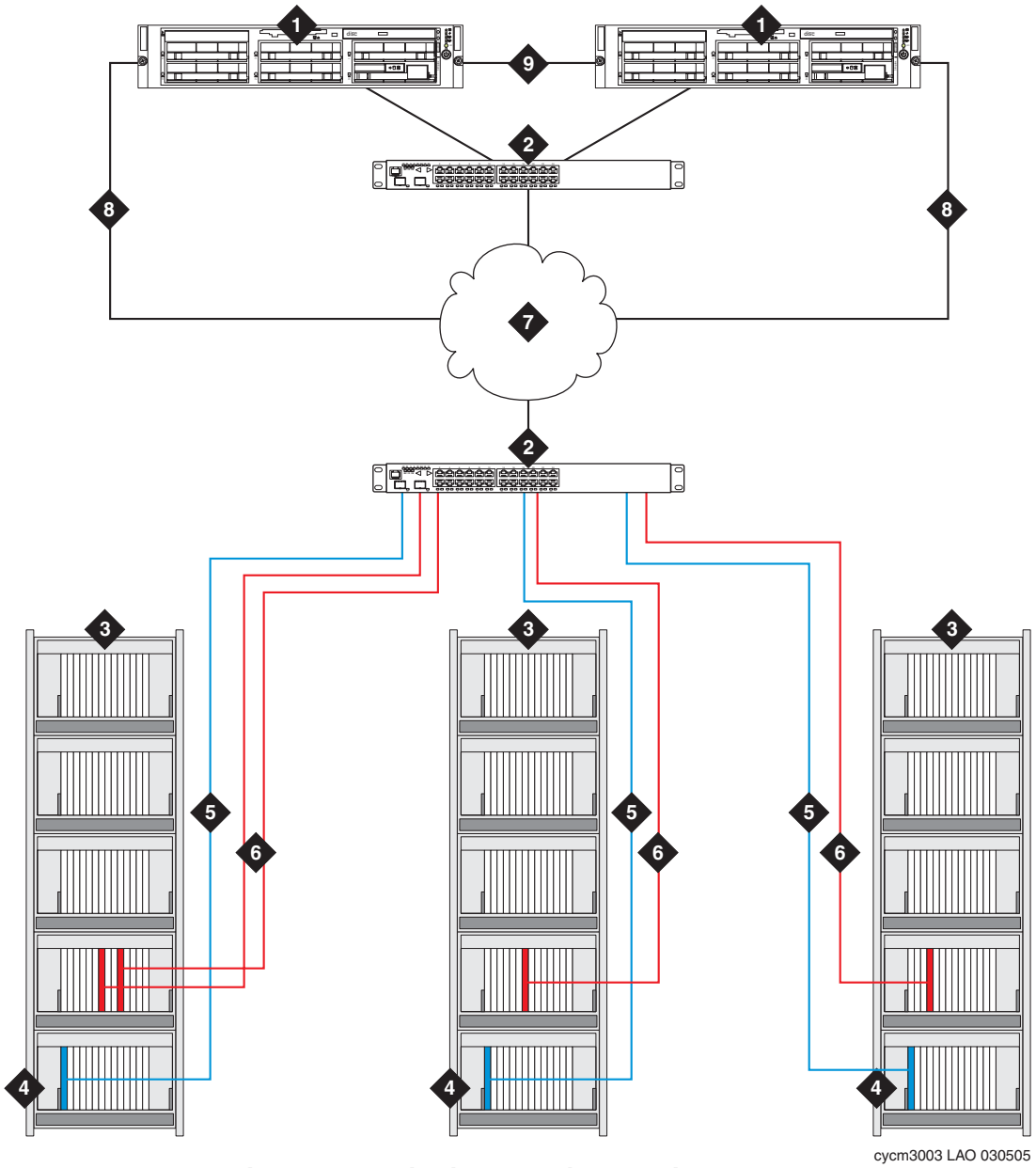

Figure notes: S8700-series IP-PNC single control network

1. S8710/S8720 Media Server
2. Ethernet Switch. For local LAN connections, the same Ethernet switch may connect both the media servers and the media gateways. For remote LAN/WAN connections, the remote gateway(s) must have an Ethernet switches at the remote location.
3. PNs (G650 Media Gateway or stack [shown in figure]). May also be a G600 or CMC1 Media Gateway or stack from an S8100 or DEFINITY Server CSI migration, an MCC1 Media Gateway from a DEFINITY Server SI or R migration, or an SCC1 Media Gateway.

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Figure notes: S8700-series IP-PNC single control network (continued)

4. PN control gateway, in the A position, which contains:

- A TN2312AP/BP IPSI circuit pack for IP connection to server.

  NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.

5. IPSI-to-server control network connection via Ethernet switch

6. LAN connections of TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints

NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, PNs, and adjunct systems. These circuit packs may be inserted into a port gateway (shown in figure) or the PN control gateway.

7. Customer LAN/WAN

8. LAN connections of media servers for remote administration

9. Duplicated server links, including the link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.

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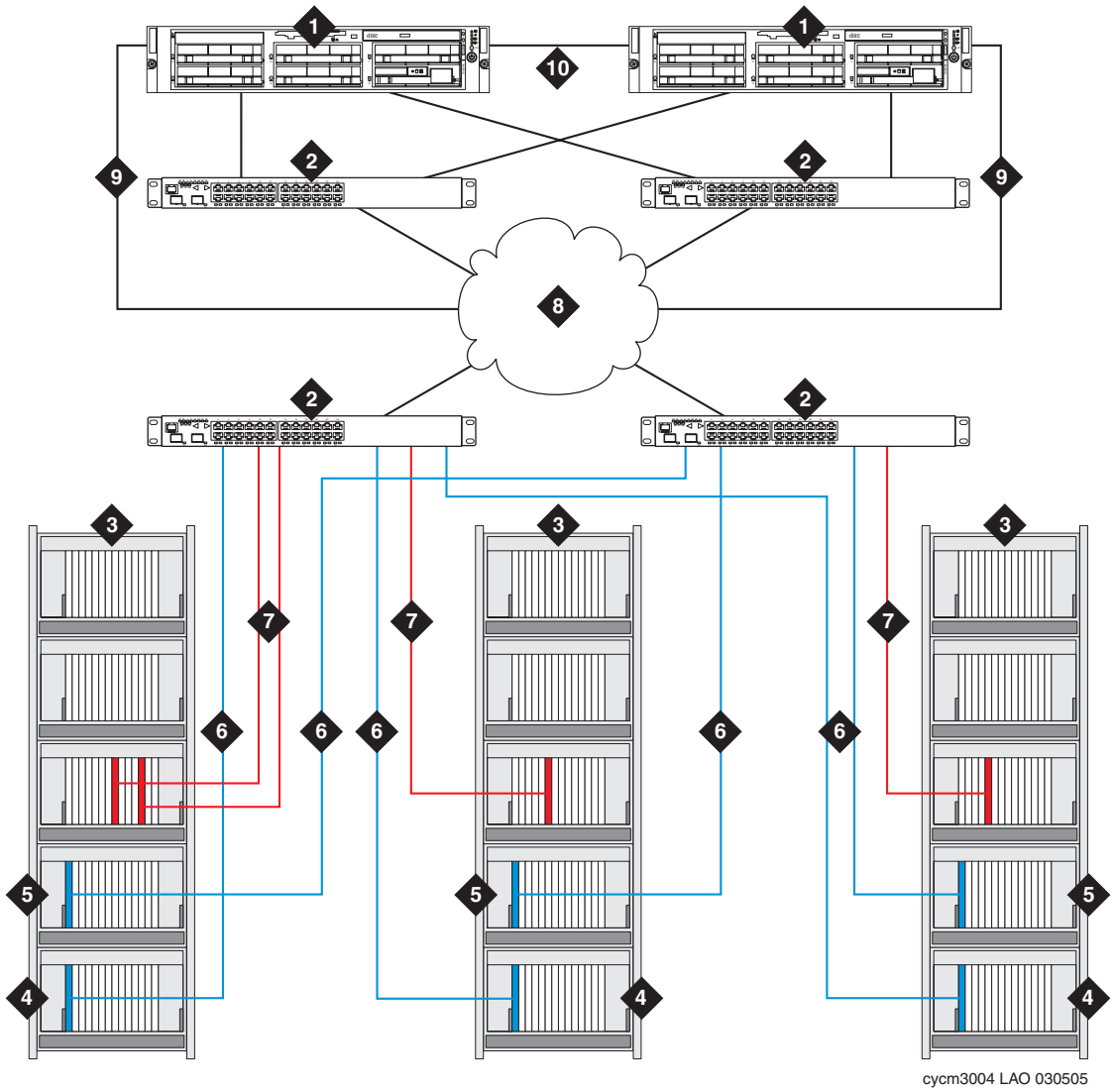
S8700-series IP-PNC (duplicated control network)

The S8700-series Media Server IP-PNC high reliability configuration is the same as the standard reliability configuration, except for the following differences:

- There are duplicated Ethernet switches, with each server connected to each Ethernet switch
- Each PN has duplicated TN2312AP/BP IPSI circuit packs. One IPSI circuit pack in each PN is connected through one Ethernet switch and the other IPSI circuit pack is connected through the other Ethernet switch

Figure 5: S8700-series IP-PNC duplicated control network



cycm3004 LAO 030505

Figure notes: S8700-series IP-PNC duplicated control network

1. S8710/S8720 Media Server
2. Ethernet Switch. For local LAN connections, the same pair of Ethernet switches may connect both the media servers and the media gateways. For remote LAN/WAN connections, the remote gateway(s) must have a pair of Ethernet switches at the remote location.
3. PNs (G650 Media Gateway or stack [shown in figure]). May also be an SCC1 stack or MCC1 Media Gateway from a DEFINITY Server SI or R migration.
4. PN control gateway, in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.

     NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.

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Figure notes: S8700-series IP-PNC duplicated control network (continued)

5. Duplicated expansion control gateway, in the B position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to control network.
6. IPSI-to-server control network connection via Ethernet switch
7. LAN connections of TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints

   NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, PNs, and adjunct systems. These circuit packs may be inserted into a port carrier (shown in figure), the PN control carrier, or the duplicated control carrier.
8. Customer LAN
9. LAN connections of media servers for remote administration
10. Duplicated server links, including the link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.

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S8700-series IP-PNC (duplicated control and duplicated bearer network)

The S8700-series Media Server IP-PNC critical reliability configuration (duplicated control and duplicated bearer network) is the same as the high reliability configuration, except for the following differences:

- Each PN has duplicated TN2602AP IP Media Resource 320 circuit packs. One TN2602 circuit pack in each PN is connected through one Ethernet switch and the other TN2602 circuit pack is connected through the other Ethernet switch.
- A TN771DP Maintenance Test circuit pack must also be installed in each PN that has duplicated control and bearer network connections.

Figure 6: S8700-series IP-PNC duplicated control and duplicated bearer network

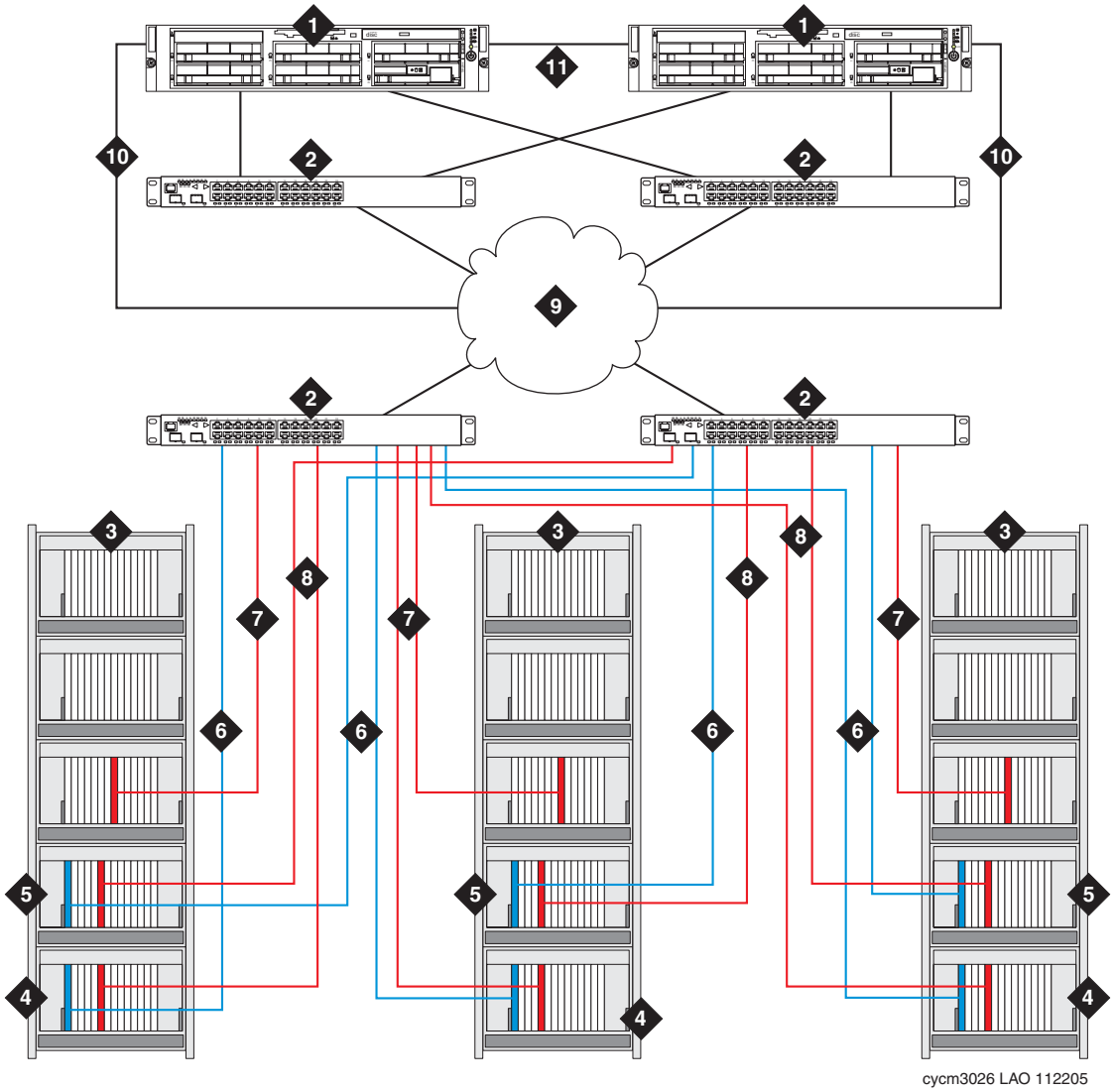

Figure notes: S8700-series IP-PNC duplicated control and duplicated bearer network

1. S8700-series Media Server
2. Ethernet Switch. For local LAN connections, the same pair of Ethernet switches may connect both the media servers and the media gateways. For remote LAN/WAN connections, the remote gateway(s) must have a pair of Ethernet switches at the remote location.
3. PNs (G650 Media Gateway or stack [shown in figure]). May also be an SCC1 stack or MCC1 Media Gateway from a DEFINITY Server SI or R migration.

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Figure notes: S8700-series IP-PNC duplicated control and duplicated bearer network (continued)

4. PN control gateway, in the A position, which contains:

- A TN2312AP/BP IPSI circuit pack for IP connection to server.

  NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.

- A TN2602AP IP Media Resource 320 for PN bearer connections over the LAN

  NOTE: The TN2602AP circuit pack may be placed in any gateway in the PN. However, the pair of TN2602 circuit packs should be separated between two different gateways whenever possible.

5. Duplicated expansion control gateway, in the B position, which contains:

- A TN2312AP/BP IPSI circuit pack for IP connection to control network.
- A TN2602AP IP Media Resource 320 for PN bearer connections over the LAN

  NOTE: The TN2602AP circuit pack may be placed in any gateway in the PN. However, the pair of TN2602 circuit packs should be separated between two different gateways whenever possible.

6. IPSI-to-server control network connection via Ethernet switch

7. LAN connection of the TN799DP C-LAN for control of IP endpoints

NOTE: The number of TN799DP circuit packs varies, depending on the number of IP endpoints, PNs, and adjunct systems. These circuit packs may be inserted into a port carrier (shown in figure), the PN control carrier, or the duplicated control carrier.

8. LAN connections of TN2602AP IP Media Resource 320 circuit packs for IP-TDM voice processing

9. Customer LAN

10. LAN connections of media servers for remote administration

11. Duplicated server links, including the link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication

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Sample S8700-series IP-PNC configuration (duplicated control and duplicated bearer)

The S8700-series Media Server IP-PNC configuration with duplicated control and duplicated bearer network is the same as the duplicated control configuration, except for the following differences:

- Each PN has duplicated TN2602AP IP Media Resource 320 circuit packs. One TN2602 circuit pack in each PN is connected through one Ethernet switch and the other TN2602 circuit pack is connected through the other Ethernet switch.
- A TN771DP maintenance test circuit pack is required each IP-PNC PN that has both duplicated control and duplicated bearer.

Figure 7: S8700-series IP-PNC duplicated control and duplicated bearer network

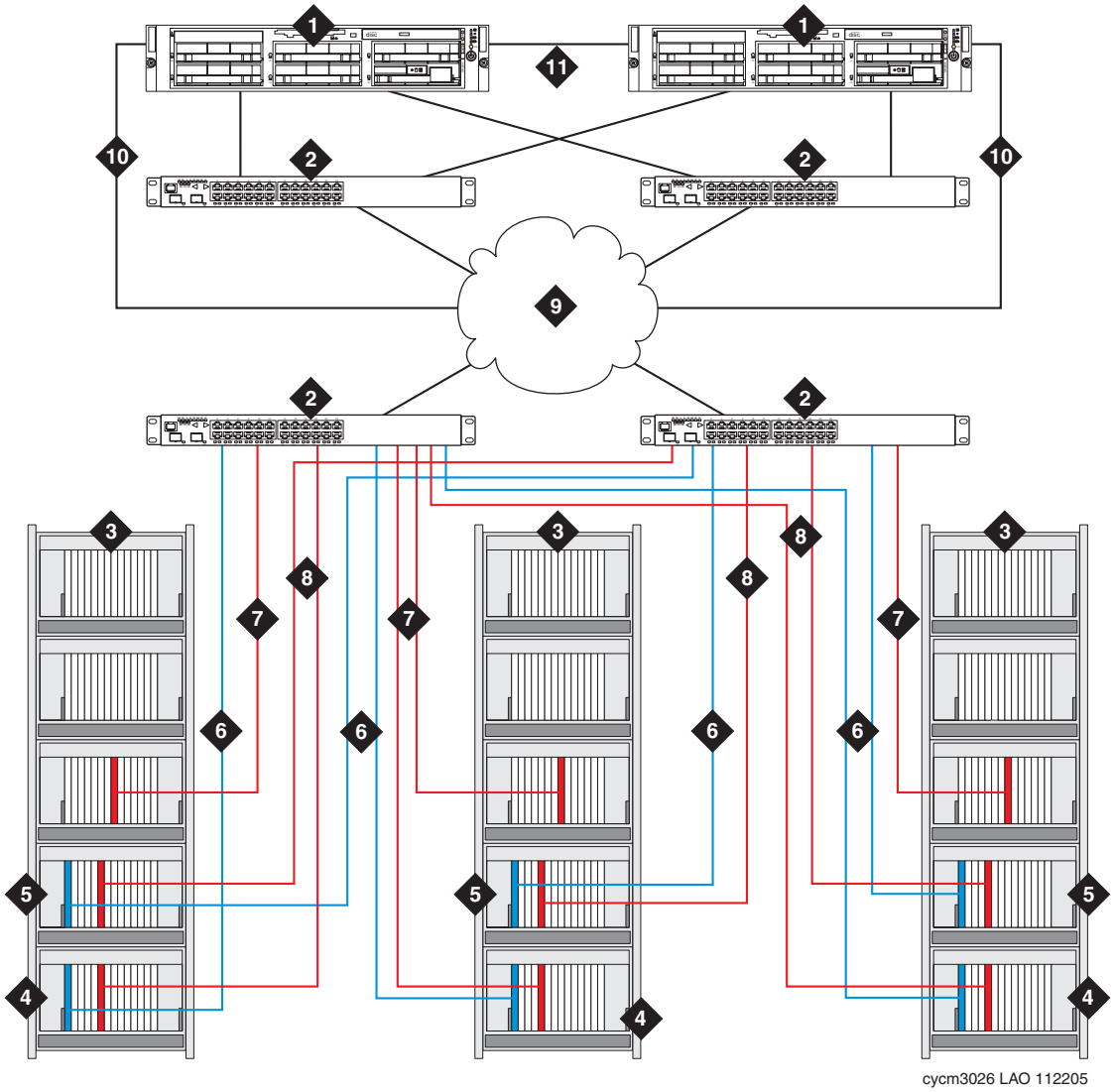

Figure notes: S8700-series IP-PNC duplicated control and duplicated bearer network

1. S8700/S8710/S8720 Media Server
2. Ethernet Switch. For local LAN connections, the same pair of Ethernet switches may connect both the media servers and the media gateways. For remote LAN/WAN connections, the remote gateway(s) must have a pair of Ethernet switches at the remote location.
3. PNs (G650 Media Gateway or stack [shown in figure]). May also be an SCC1 stack or MCC1 Media Gateway from a DEFINITY Server SI or R migration.

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Figure notes: S8700-series IP-PNC duplicated control and duplicated bearer network (continued)

4. PN control gateway, in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.

     NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.
   - A TN2602AP IP Media Resource 320 for PN bearer connections over the LAN

     NOTE: The TN2602AP circuit pack may be placed in any gateway in the PN. However, the pair of TN2602 circuit packs should be separated between two different gateways whenever possible.
5. Duplicated expansion control gateway, in the B position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to control network.
   - A TN2602AP IP Media Resource 320 for PN bearer connections over the LAN

     NOTE: The TN2602AP circuit pack may be placed in any gateway in the PN. However, the pair of TN2602 circuit packs should be separated between two different gateways whenever possible.
6. IPSI-to-server control network connection via Ethernet switch
7. LAN connection of the TN799DP C-LAN for control of IP endpoints

   NOTE: The number of TN799DP circuit packs varies, depending on the number of IP endpoints, PNs, and adjunct systems. These circuit packs may be inserted into a port carrier (shown in figure), the PN control carrier, or the duplicated control carrier.
8. LAN connections of TN2602AP IP Media Resource 320 circuit packs for IP-TDM voice processing
9. Customer LAN
10. LAN connections of media servers for remote administration
11. Duplicated server links, including the link for translations transfer and the link for control data sharing

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S8700-series direct-connect (single control network)

In this configuration, one PN is connected to the server over an Ethernet connection. Fiber links connect up to two additional PNs to each other. This configuration also requires either a dual-NIC card in the S8700-series Media Server or an interim Ethernet switch.

IPSI-connected PN

Only the G650 media gateway is available for new installations. However, because different migrations from older systems are supported, the PN connected to the S8700-series Media Server can consist of one of three gateways:

- G650 media gateway

  A G650 PN can consist of one to five G650 gateways in a stack connected by a TDM/LAN bus cable. One gateway, serving as control gateway in position A at the bottom of the stack, contains the following:

  - TN2312BP IPSI circuit pack

- SCC1 media gateway

  An SCC1 PN can consist of one to four SCC1 gateways in a stack connected by a TDM/LAN bus cable. One gateway, serving as control gateway in position A at the bottom of the stack, contains the following:

  - TN2312AP/BP IPSI circuit pack

  Note:

  The TN2404 and TN2401 processor circuit packs, needed for the SI model, are *not* used and must be removed if the SCC1 is being migrated from a DEFINITY server.

- MCC1 media gateway

  An MCC1 PN has from one to five carriers in an MCC1 gateway connected by a TDM/LAN bus cables. One carrier, serving as control carrier in position A in the middle of the stack, contains the following:

  - TN2312AP/BP IPSI circuit pack

  Note:

  The control carrier for a DEFINTY Server SI or R is not used and must be removed and replaced with an expansion control carrier if the MCC1 is being migrated. The processor circuit packs, needed for the SI or R models, are not used and must be removed. Other PNs can also be MCC1 Gateways.

PNs not IPSI-connected

In a S8700-series Media Server direct connect configuration, additional PNs (up to two only) may be connected to the IPSI-connected PN using fiber optic cable between external interface (EI) TN570B (version 7 or later) circuit packs. The cables are connected to the circuit packs using short-range or long-range multi-mode transceivers, or single-mode transceivers, depending on the distance between PNs.

The TN570B circuit packs reside in the control carrier (MCC1) or control gateway (G650 or SCC1) of each PN. The control carrier or gateway in each additional PN also must contain a TN2182 Tone Clock circuit pack (SCC1 or MCC1) or a maintenance-only TN2312ABP IPSI circuit pack (G650).

Note:

Straight fiber connections between TN570B circuit packs may be up to 200 feet (61 meters) (see TN570B Expansion Interface PN connections up to 200 ft. on page 75). If the distance between PNs is greater, Light guide interface units (LIUs) must also be used to connect the fiber cables or the connection must use TN1654 DS1 converters. Lengths of fiber, including connections through LIUs or DS1 converters, are:

- 4900 feet (1493.5 meters) (see TN570B Expansion Interface PN connections up to 4900/25000 ft. and 22 miles. on page 76)

- 25,000 feet (7620 meters) in multimode (see TN570B Expansion Interface PN connections up to 4900/25000 ft. and 22 miles. on page 76)
- 21.7 miles (34.9 kilometers) in single mode (see TN570B Expansion Interface PN connections up to 4900/25000 ft. and 22 miles. on page 76)
- 200 miles (322 kilometers) (see TN1654 DS1 Converter/TN570B Expansion Interface PN connections up to 200 miles. on page 78)

Note:

You *cannot* connect additional PNs that contain CMC1 or G600 Media Gateways.

TN2602AP circuit packs for duplicated bearer

For an S8700-series Media Server, any individual fiber-PNC PN can contain load-balancing or duplicated TN2602AP circuit packs. However, TN2602AP circuit packs do *not* need to be implemented uniformly within the system. Thus, some PNs may have no TN2602AP circuit pack, some PNs may have load-balancing TN2602AP circuit packs, and some PNs may have duplicated TN2602AP circuit packs. Thus, an S8700-series Media Server can have duplicated bearer connections, even though it does not support duplicated control or fiber-based duplicated bearer.

Figure 8: S8700-series direct-connect single control network

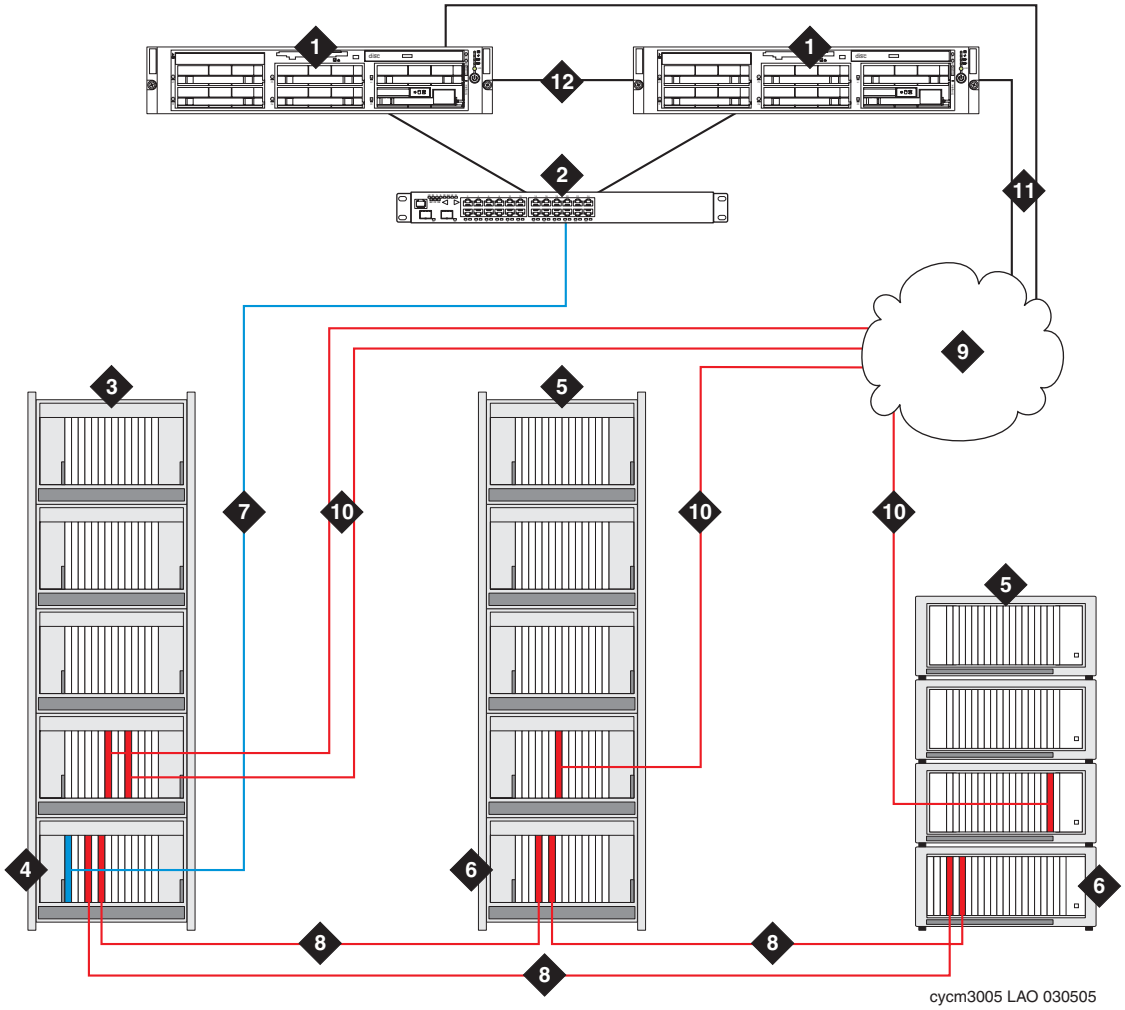

Figure notes: S8700-series direct-connect single control network

1. S8700-series Media Server
2. Ethernet Switch
3. Direct-connect PN (G650 Media Gateway stack [shown in figure],MCC1 Media Gateway, or SCC1 Media Gateway stack [shown in figure], consisting of at least two media gateways or carriers).
4. Media Gateway (G650) or expansion port network (EPN) control gateway (SCC1) or carrier (MCC1), in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.

     NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.

   - Two TN570B EI circuit packs for bearer and control network connections to the other two PNs (if any).
5. PN (G650 Media Gateway or G650 stack [shown in figure], MCC1 Media Gateway, or SCC1 Media Gateway or SCC1 stack).

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Figure notes: S8700-series direct-connect single control network (continued)

6. PN control gateway or carrier, in the A position, which contains two TN570B EI circuit packs for bearer and control network connections to the other two PNs.

   NOTE: One TN2182 Tone Clock circuit pack must also be present per PN if the PN(s) consist of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312AP/BP IPSI circuit pack must be present per PN if the PN(s) consist of G650 Media Gateways.

7. IPSI-to-server control network connection via Ethernet switch

8. TN 570/570 fiber connections between PNs

9. Customer LAN

10. LAN connections, if any, of optional TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints

    NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, PNs, and adjunct systems. These circuit packs are optional for PNs in a direct-connect network and may be inserted into a port carrier (shown in figure) or the PN control carrier. However, the C-LAN circuit pack is required for downloads of firmware updates.

11. LAN connections of media servers for remote administration

12. Duplicated server links, including the link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.

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S8700-series direct-connect (duplicated control network)

For high reliability in a direct-connect configuration, the control network is duplicated. This configuration is basically the same as that of the single control network configuration, except that a second carrier or gateway is added in the B position to provide a second IPSI connection to the servers. In this case, the normally-active server is connected to the control carrier/ gateway IPSI circuit pack, and the standby server is connected to the second carrier/gateway IPSI circuit pack. See S8700-series direct-connect duplicated control network on page 47.

All other connections between the PNs are the same.

Figure 9: S8700-series direct-connect duplicated control network

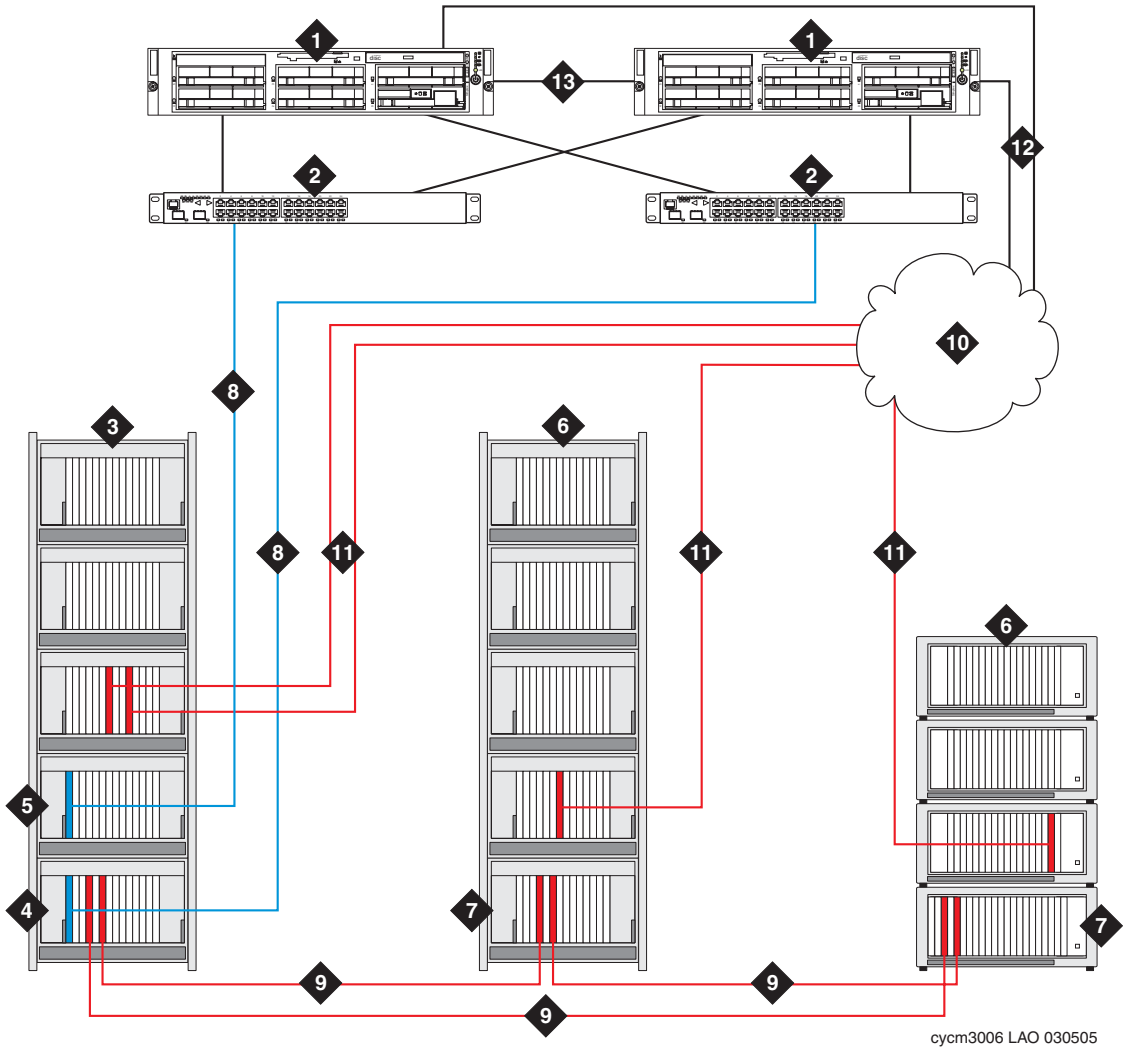

Figure notes: S8700-series direct-connect duplicated control network

1. S8700-series Media Server
2. Ethernet Switch
3. Direct-connect PN (G650 Media Gateway stack [shown in figure], MCC1 Media Gateway, or SCC1 Media Gateway stack), consisting of at least two media gateways or carriers.
4. PN control gateway or carrier, in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.

     NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.
   - Two TN570B EI circuit packs for bearer and control network connections to the other two PNs (if any).

1 of 2

Figure notes: S8700-series direct-connect duplicated control network (continued)

5. Duplicated expansion control gateway or carrier, in the B position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to control network
6. PN (G650 Media Gateway stack [shown in figure], MCC1 Media Gateway, SCC1 Media Gateway stack [shown in figure]), consisting of at least two media gateways or carriers.
7. PN control gateway or carrier, which contains two TN570B EI circuit packs for bearer and control network connections to the other two PNs.

   NOTE: One TN2182 Tone Clock circuit pack must also be present per PN if the PN(s) consist of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312BP IPSI circuit pack must be present per PN if the PN(s) consist of G650 Media Gateways.

   The control gateway or carrier is always in the A position in the MCC1 or gateway stack.
8. IPSI-to-server control network connection via Ethernet switch
9. TN 570/570 fiber connections between PNs
10. Customer LAN
11. LAN connections, if any, of optional TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints

    NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, PNs, and adjunct systems. These circuit packs are optional for PNs in a direct-connect network and may be inserted into a port carrier (shown in figure), the PN control carrier, or the duplicated control carrier. However, the C-LAN circuit pack is required for downloads of firmware updates.
12. LAN connections of media servers for remote administration
13. Duplicated server links, including the link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.

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S8700-series direct-connect (duplicated control and bearer networks)

For critical reliability in a direct-connect configuration, both the control network and bearer network are duplicated. This configuration is basically the same as the duplicated-control-network-only (high reliability) configuration, except that a second carrier or gateway is added in each additional PN with optic fiber link connections to the second carrier or gateway of the IPSI-connect PN. See S8700-series direct-connect duplicated control network and duplicated voice-bearer network on page 49.

All other connections between the PNs are the same as those of the duplicated-control-network-only configuration.

Figure 10: S8700-series direct-connect duplicated control network and duplicated voice-bearer network

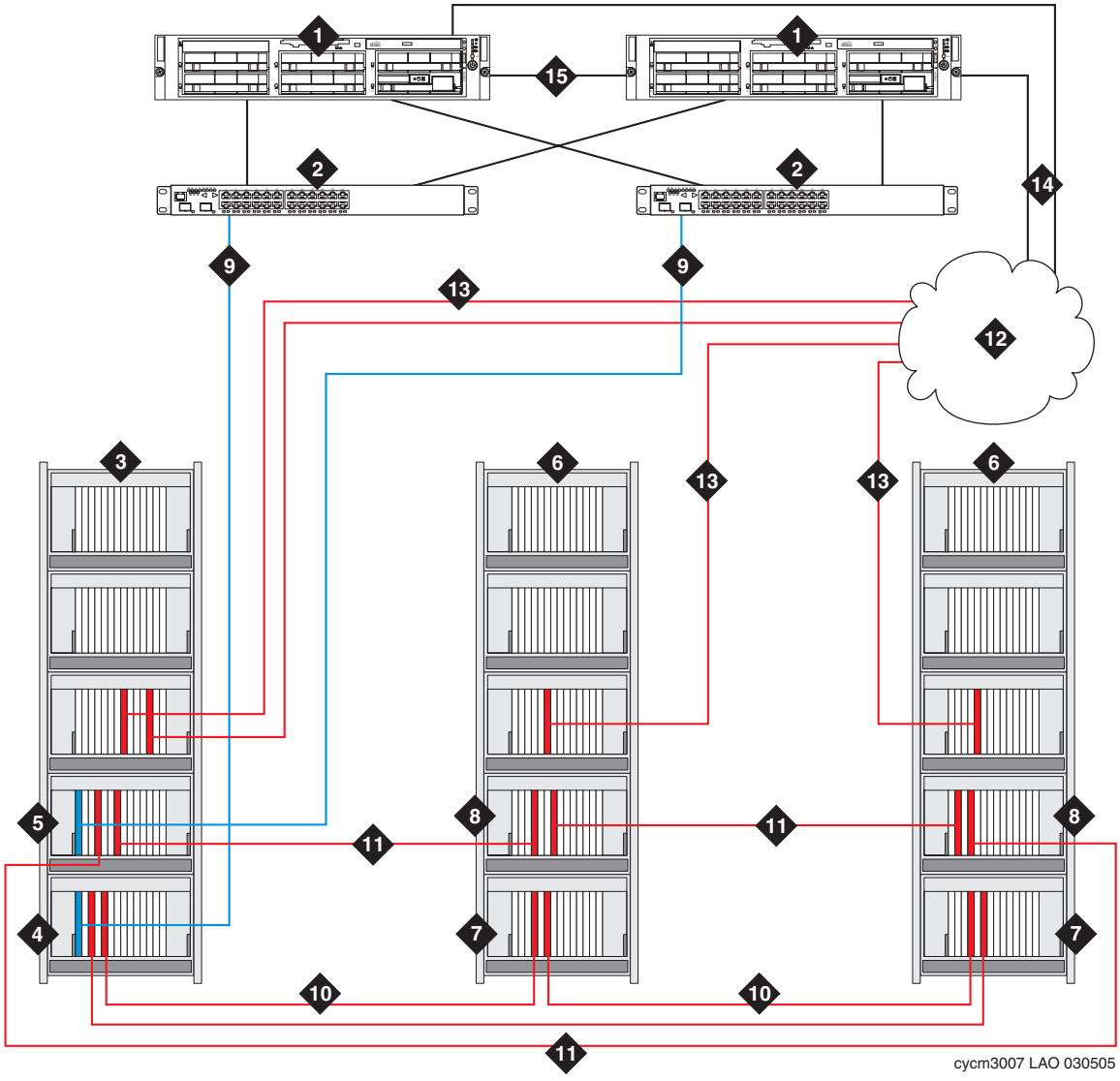

Figure notes: S8700-series direct-connect duplicated control network and duplicated voice-bearer network

1. S8700-series Media Server
2. Ethernet Switch
3. Direct-connect PN (G650 Media Gateway stack [shown in figure],MCC1 Media Gateway, or SCC1 Media Gateway stack, consisting of at least two media gateways or carriers).

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Figure notes: S8700-series direct-connect duplicated control network and duplicated voice-bearer network (continued)

4. PN control gateway or carrier, in the A position, which contains:

- A TN2312AP/BP IPSI circuit pack for IP connection to server.

  NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.

- Two TN570B EI circuit packs for bearer and control network connections to the other two PNs (if any).

5. Duplicated expansion control cabinet or carrier, in the B position, which contains:

- A TN2312AP/BP IPSI circuit pack for IP connection to control network.
- Two TN570B EI circuit packs for bearer and control network connections to the other two PNs.

6. PN (G650 Media Gateway stack (shown in figure), MCC1 Media Gateway, SCC1 Media Gateway stack), consisting of at least two media gateways or carriers.

7. PN control gateway or carrier, in the A position, which contains two TN570B EI circuit packs for bearer and control network connections to the other two PNs.

NOTE: One TN2182 Tone Clock circuit pack must also be present per PN if the PN(s) consist of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312AP/BP IPSI circuit pack must be present per PN if the PN(s) consist of G650 Media Gateways.

8. Duplicated expansion control cabinet or carrier, in the B position, which contains:

- Two TN570B EI circuit packs for bearer and control network connections to the other two PNs.

9. IPSI-to-server control network connection via Ethernet switch

10. TN 570/570 fiber connections between PNs

11. Duplicated TN 570/570 fiber connections between PNs

12. Customer LAN

13. LAN connections, if any, of optional TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints

NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, PNs, and adjunct systems. These circuit packs are optional for PNs in a direct-connect network and may be inserted into a port carrier (shown in figure), the PN control carrier, or the duplicated control carrier. However, the C-LAN circuit pack is required for downloads of firmware updates.

14. LAN connections of media servers for remote administration

15. Duplicated server links, including the link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.

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Rules for TN570B circuit pack placement with SCC1/MCC1 Media Gateways

Fiber-PNC MCC1 and SCC1 Media Gateways have the following rules on the placement of TN570B External Interface circuit packs in direct connect configurations.

With a single and duplicated control network

For a direct connect configuration with a single (standard reliability) or duplicated control (high reliability) network, the placement rules are as follows:

- The IPSI-connected PN houses up to two TN570B circuit packs, the first in the A01 slot and the second in the A02 slot. These circuit packs connect to the TN570B circuit packs residing in the A01 slots only of the non-IPSI-connected PNs (up to two).
- With three PNs in the direct connect configuration, the non-IPSI-connected PNs connect over fiber to each other with a TN570B circuit pack that resides in the A02 slot in each PN.

With duplicated bearer network

For a direct connect configuration with duplicated control and a duplicated bearer (critical reliability) network, the rules for a single/duplicated control network still apply. In addition, the following rules apply for the TN570B circuit packs in the B cabinets/carriers:

- The IPSI-connected PN houses up to two TN570B circuit packs in the duplicated control cabinet/carrier, the first in the B02 slot and the second in the B03 slot. These circuit packs connect to the TN570B circuit packs residing in the B02 slots only of the non-IPSI-connected PNs (up to two).
- With three PNs in the direct connect configuration, the non-IPSI-connected PNs connect over fiber to each other with a TN570B circuit pack that resides in the B03 slot in each PN.

The following table illustrates the exact TN570B-to-TN570B connections and the required placement of the TN570B circuit packs in the PNs.

Table 1: Slot positions of connected TN570B circuit packs in SCC1/MCC1 direct connect configurations (single and duplicated control networks)

		With PN1 IPSI-connected	With PN2 IPSI-connected	With PN3 IPSI-connected
Slot Positions of connected TN570Bs[1]	Single control (A position)	1A01 connects to 2A01	2A01 connects to 1A01	3A01 connects to 1A01
		1A02 connects to 3A01	2A02 connects to 3A01	3A02 connects to 2A01
		2A02 connects to 3A02	1A02 connects to 3A02	1A02 connects to 2A02
	Duplicated control (B position)	1B02 connects to 2B02	2B02 connects to 1B02	3B02 connects to 1B02
		1B02 connects to 3B02	2B03 connects to 3B02	3B03 connects to 2B02
		2B03 connects to 3B03	1B03 connects to 3B03	1B03 connects to 2B03

1. Slot positions are in the form *uuccss*, where *uu* is the port network number, *cc* is the cabinet or carrier, and *ss* is the slot number.

Implications for migrations from DEFINITY R and SI Servers - In a migration from a DEFINITY Server R or SI to an S8700-series Media Server with a single or duplicated control network, one of the port networks in the new configuration must have either one or two IPSIs installed in the PN for connections to the server. It is recommended that the IPSI be installed in the converted processor port network (PPN) of the DEFINITY system because the TN570B fiber connections can remain as they were prior to the migration. If the IPSI is installed in a converted EPN instead, the fiber connections between the TN570Bs must be changed. Table 1 illustrates the necessary changes, assuming that PN1 represents the converted PPN.

Likewise, if the migrated configuration includes a duplicated bearer network, the recommendation to install the IPSIs in the converted PPN becomes even more compelling, If the PPN does *not* become the IPSI-connected PN, then changes to fiber connections between the TN570Bs are necessary in both the A and B positions.

Note:

The G650 Media Gateway does not restrict where the TN570B EI circuit packs are placed, except that they cannot be inserted into the A01 and B01 slots.

S8700-series Center Stage Switch (single control network)

The Center Stage Switch (CSS) is an MCC1 Media Gateway that contains a switch node carrier (SNC) in the bottom E position. The SNC, in turn, houses TN573B switch node interface (SNI) circuit packs, which connect to PNs over optic fiber cable to TN570B EI circuit packs in the PNs. A single SNC allows 15 PNs to be connected with fiber to the IPSI-connected PN. In large configurations, a second or third MCC1 may be equipped with an SNC. The SNC expansion of port networks, therefore, is as follows:

- One SN can expand to up to 15 PNs.
- Two SNs can expand to up to 29 PNs.
- Three SNs can expand to up to 44 PNs.

Note:

The fiber link connections in an S8700-series CSS configuration follow the same distance rules as those of the S8700-series direct-connect configurations.

A single IPSI circuit pack allows the server to control up to 5 PNs only. Therefore, in a configuration with many PNs, multiple PNs may be IPSI-connected.

PN configurations

The MCC1 Media Gateway with a CSS is a IPSI-connected PN that also houses an SNC. However, the MCC1 as a CSS can also be configured to house *only* the SNC, with no control or port carriers. In this case, the MCC1 connects to a IPSI-connected PN using the SNI-to-TN570B fiber connection. Therefore, in a CSS configuration, the PNs can be any of the following:

- MCC1 Media Gateway

  IPSI-connected. An MCC1 PN that is connected to the server has the same configuration as that of a IPSI-connected PN in a direct-connect with duplex-servers-only configuration.

  However, if the MCC1 PN also contains an SNC, the IPSI-connected expansion control carrier of the MCC1 must also be connected to the SNC with optic fiber from a TN570B circuit pack. Also, only 4 carriers are then available for control and port circuit packs.

  Non-IPSI-connected. An MCC1 PN that is *not* connected directly to the server has the same configuration as that of an additional MCC1 PN in a direct-connect with single control network configuration.

- G650 media gateway

  IPSI-connected. A G650 PN can consist of one or more G650 gateways in a stack (up to 5 in a stack connected by TDM/LAN bus cables). A G650 PN that is connected to the server has the same configuration as that of a IPSI-connected PN in a direct-connect with single control network configuration.

  Not IPSI-connected. A G650 PN that is *not* connected directly to the server but is connected to the SNC has the same configuration as that of an additional G650 PN in a direct-connect with single control network configuration.

- SCC1 media gateway

  IPSI-connected. An SCC1 PN can consist of one or more SCC1 gateways in a stack (up to 4 in stack connected by TDM/LAN bus cables). An SCC1 PN that is connected to the server has the same configuration as that of a IPSI-connected PN in a direct-connect with single control network configuration.

  Not IPSI-connected. An SCC1 PN that is *not* connected directly to the server but is connected to the SNC has the same configuration as that of an additional SCC1 PN in a direct-connect with single control network configuration.

  Note:

  With the S8700-series Media Server, the SNC is *not* connected to the other carriers in the MCC1 cabinet with TDM/LAN bus cables.

In the following example, 5 PNs, one of which is embedded in the MCC1 with the CSS, requires two IPSIs.

TN2602AP circuit packs for duplicated bearer

For an S8700-series Media Server, any individual fiber-PNC PN can contain load-balancing or duplicated TN2602AP circuit packs. However, TN2602AP circuit packs do *not* need to be implemented uniformly within the system. Thus, some PNs may have no TN2602AP circuit pack, some PNs may have load-balancing TN2602AP circuit packs, and some PNs may have duplicated TN2602AP circuit packs. Thus, an S8700-series Media Server can have duplicated bearer connections, even though it does not support duplicated control or fiber-based duplicated bearer.

Figure 11: S8700-series Center Stage Switch single control network

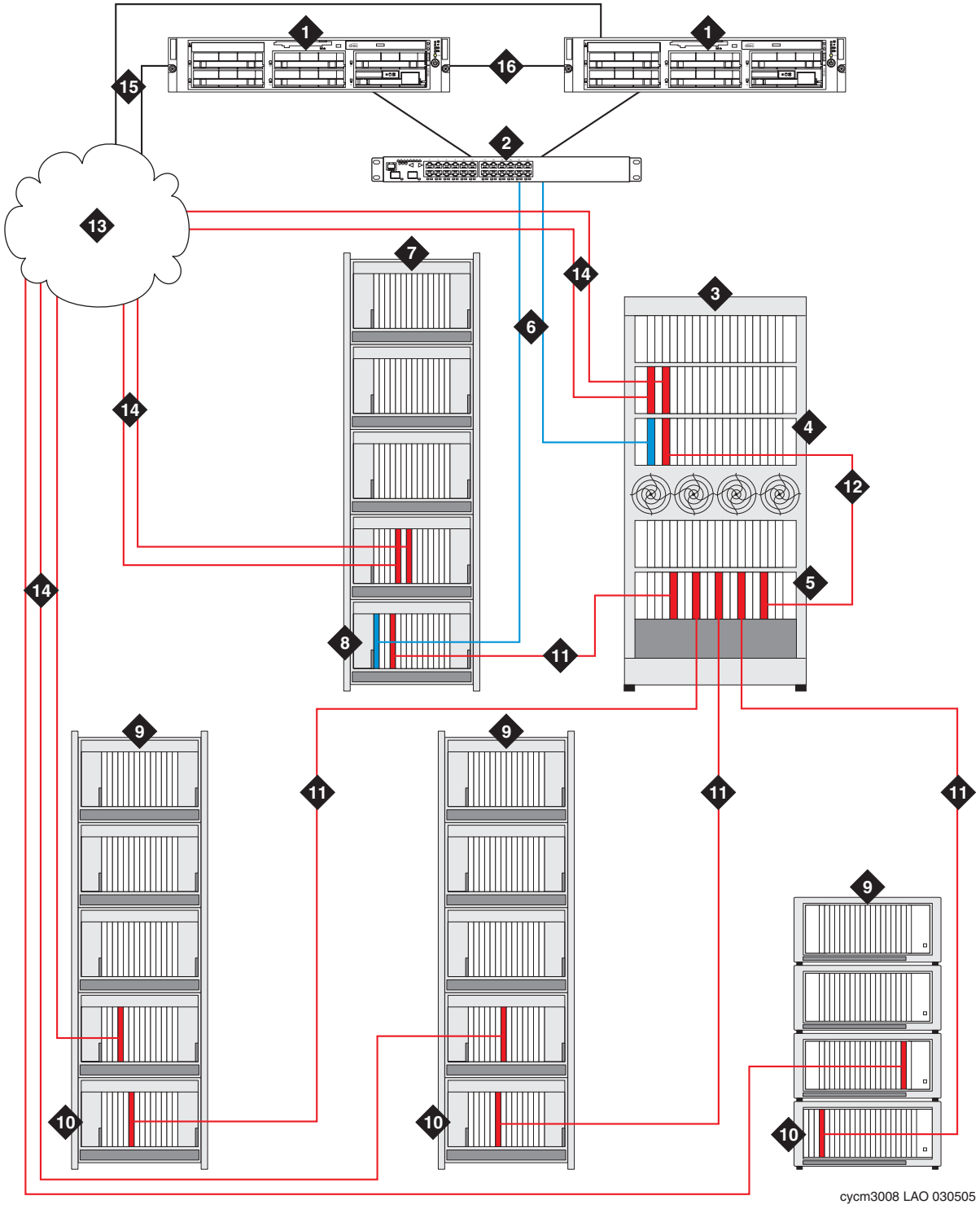

Figure notes: S8700-series Center Stage Switch single control network

1. S8700-series Media Server
2. Ethernet Switch
3. MCC1 Media Gateway (CSS and PN)
4. PN control carrier, in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.
   - A TN570B EI circuit pack for bearer and control network connections to the Switch Node Carrier (SNC).
5. SNC, in the E position, which contains:
   - Multiple TN573B SNI circuit packs for EI connections to PNs
6. IPSI-to-server control network connection via Ethernet switch
7. IPSI-connected PN (G650 Media Gateway or stack [shown in figure],MCC1 Media Gateway, or SCC1 Media Gateway stack).
8. PN control gateway or carrier, in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.

     NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.
   - A TN570B EI circuit pack for bearer and control network connections to the SNI.
9. PN (MCC1 Media Gateway, SCC1 Media Gateway stack [shown in figure], or G650 Media Gateway stack [shown in figure]) consisting of one or more media gateways or carriers.
10. PN control gateway or carrier, in the A position, which contains:
    - A TN570B EI circuit pack for bearer and control network connections to the SNI.

      NOTE: One TN2182 Tone Clock circuit pack must also be present per PN if the PN(s) consist of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312BP IPSI circuit pack must be present per PN if the PN(s) consist of G650 Media Gateways.
11. TN 570B/573B fiber connections between PNs and SNC
12. TN 573B/570B fiber connections between the SNC and the MCC1's A carrier (if the MCC1 is a PN)
13. Customer LAN
14. LAN connections, if any, of optional TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints

    NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, PNs, and adjunct systems. These circuit packs are optional for PNs in a CSS-connected network and may be inserted into a port carrier (shown in figure), the PN control carrier, or the duplicated control carrier. However, the C-LAN circuit pack is required for downloads of firmware updates.
15. LAN connections of media servers for remote administration
16. Duplicated server links, including the link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.

S8700-series Center Stage Switch (duplicated control network)

For high reliability in a CSS configuration, the control network is duplicated. This configuration is basically the same as that of the single control network configuration, except that a second carrier or gateway is added in the B position of each IPSI-connected PN to provide a second IPSI connection to the servers. In addition, this configuration contains duplicated Ethernet switches, each connected to both S8700-series Media Servers.

Note:

With the S8700-series Media Server, the SNC is *not* connected to the other carriers in the MCC1 cabinet with TDM/LAN bus cables.

IPSI-connected PNs

Because a single IPSI circuit pack allows the server to control up to 5 PNs only. a configuration with many PNs and duplicated control networks can require many IPSI circuit packs.

TN2602AP circuit packs for duplicated bearer

For an S8700-series Media Server, any individual fiber-PNC PN can contain load-balancing or duplicated TN2602AP circuit packs. However, TN2602AP circuit packs do *not* need to be implemented uniformly within the system. Thus, some PNs may have no TN2602AP circuit pack, some PNs may have load-balancing TN2602AP circuit packs, and some PNs may have duplicated TN2602AP circuit packs. Thus, an S8700-series Media Server can have duplicated bearer connections, even though it does not support duplicated control or fiber-based duplicated bearer.

Figure 12: S8700-series Center Stage Switch duplicated control networks

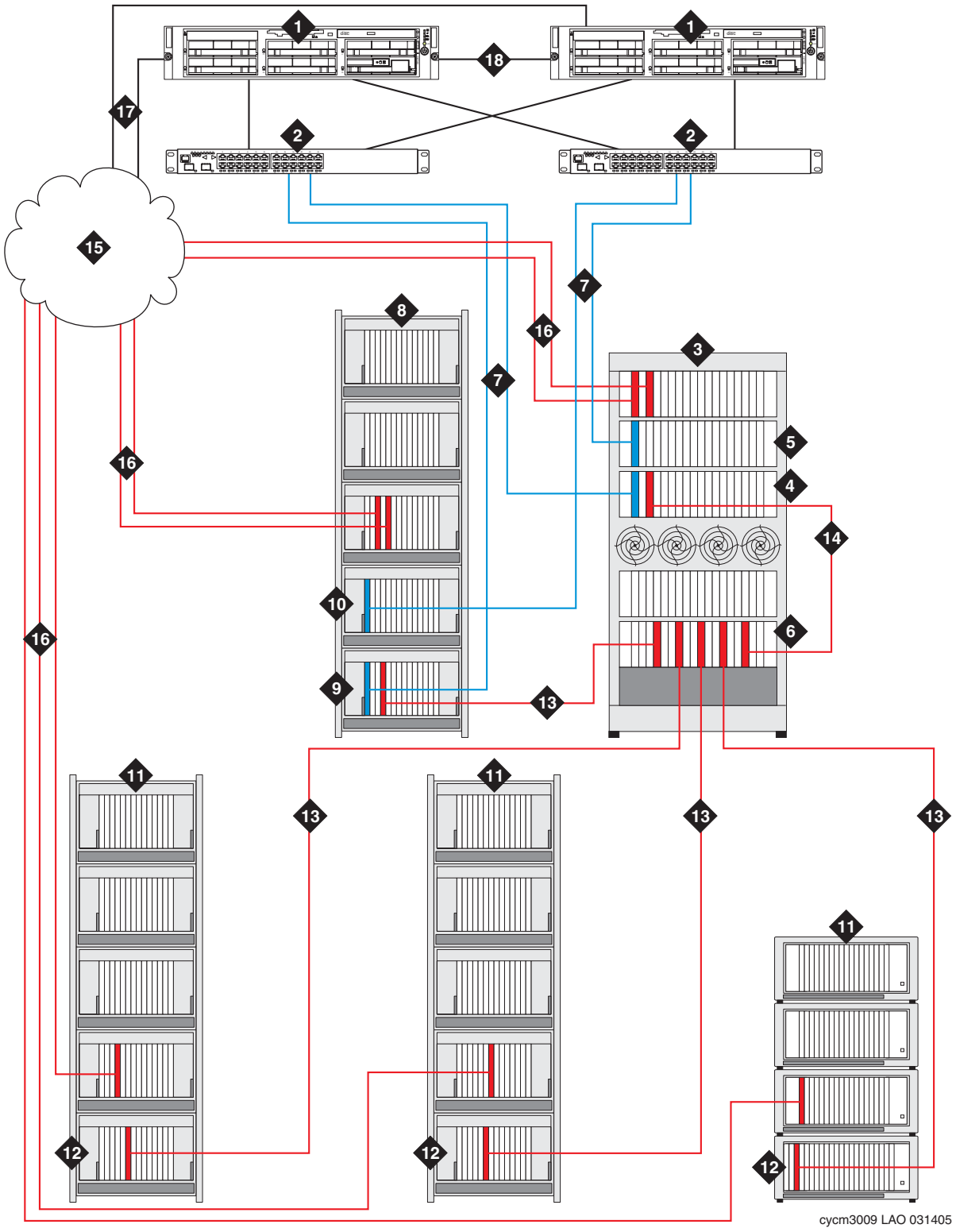

Figure notes: S8700-series Center Stage Switch duplicated control networks

1. S8700-series Media Server
2. Ethernet Switch
3. MCC1 Media Gateway (CSS and PN)
4. PN control carrier, in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.
   - A TN570B EI circuit pack for bearer and control network connections to the Switch Node Carrier (SNC).
5. Duplicated control carrier, in the B position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to duplicated control network.
6. SNC, in the E position, which contains:
   - Multiple TN573B SNI circuit packs for EI connections to PNs
7. Dedicated IPSI-to-server control network connection via Ethernet switch
8. IPSI-connected PN (G650 Media Gateway stack [shown in figure],MCC1 Media Gateway, or SCC1 Media Gateway stack, consisting of at least two media gateways or carriers).
9. PN control gateway or carrier, in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.

     NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.
   - A TN570B EI circuit pack for bearer and control network connections to the SNC.
10. Duplicated control gateway, in the B position, which contains:
    - A TN2312AP/BP IPSI circuit pack for IP connection to server.
11. Fiber-PNC PN (G650 Media Gateway stack [shown in figure], MCC1 Media Gateway, SCC1 Media Gateway stack [shown in figure]), consisting of at least two media gateways or carriers.
12. PN control gateway or carrier, in the A position which contains:
    - A TN570B EI circuit pack for bearer and control network connections to the SNI.

      NOTE: One TN2182 Tone Clock circuit pack must also be present per PN if the PN(s) consist of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312BP IPSI circuit pack must be present per PN if the PN(s) consist of G650 Media Gateways.
13. TN 570B/573B fiber connections between PNs and SNC
14. TN 573B/570B fiber connections between the SNC and the MCC1's A carrier (if the MCC1 is a PN)
15. Customer LAN
16. LAN connections, if any, of optional TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints

    NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, PNs, and adjunct systems. These circuit packs are optional for PNs in a CSS-connected network and may be inserted into a port carrier (shown in figure), the PN control carrier, or the duplicated control carrier. However, the C-LAN circuit pack is required for downloads of firmware updates.
17. LAN connections of media servers for remote administration
18. Duplicated server links, including the link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.

S8700-series Center Stage Switch (duplicated control and bearer networks)

Like the high reliability CSS configuration, the critical reliability CSS configuration duplicates the control network between the servers and the PNs. In addition, this configuration contains duplicated switch node carriers in each CSS, which duplicates the bearer network between PNs. Each PN, in turn, contains duplicated TN570B external interface circuit packs that connect to both switch node carriers. In addition, each non-IPSI-connected PN must have duplicate TN2182CTone Clock circuit packs. And finally, in each location of a PN or group of PNs, one of the PNs must have a TN771 Maintenance Test circuit pack.

Note:

With the S8700-series Media Server, the SNCs are *not* connected to the other carriers in the MCC1 cabinet with TDM/LAN bus cables.

TN2602AP circuit packs for duplicated bearer

For an S8700-series Media Server, any individual fiber-PNC PN can contain load-balancing or duplicated TN2602AP circuit packs. However, TN2602AP circuit packs do *not* need to be implemented uniformly within the system. Thus, some PNs may have no TN2602AP circuit pack, some PNs may have load-balancing TN2602AP circuit packs, and some PNs may have duplicated TN2602AP circuit packs. Thus, an S8700-series Media Server can have duplicated bearer connections, even though it does not support duplicated control or fiber-based duplicated bearer.

Figure 13: S8700-series Center Stage Switch duplicated control and duplicated voice-bearer networks

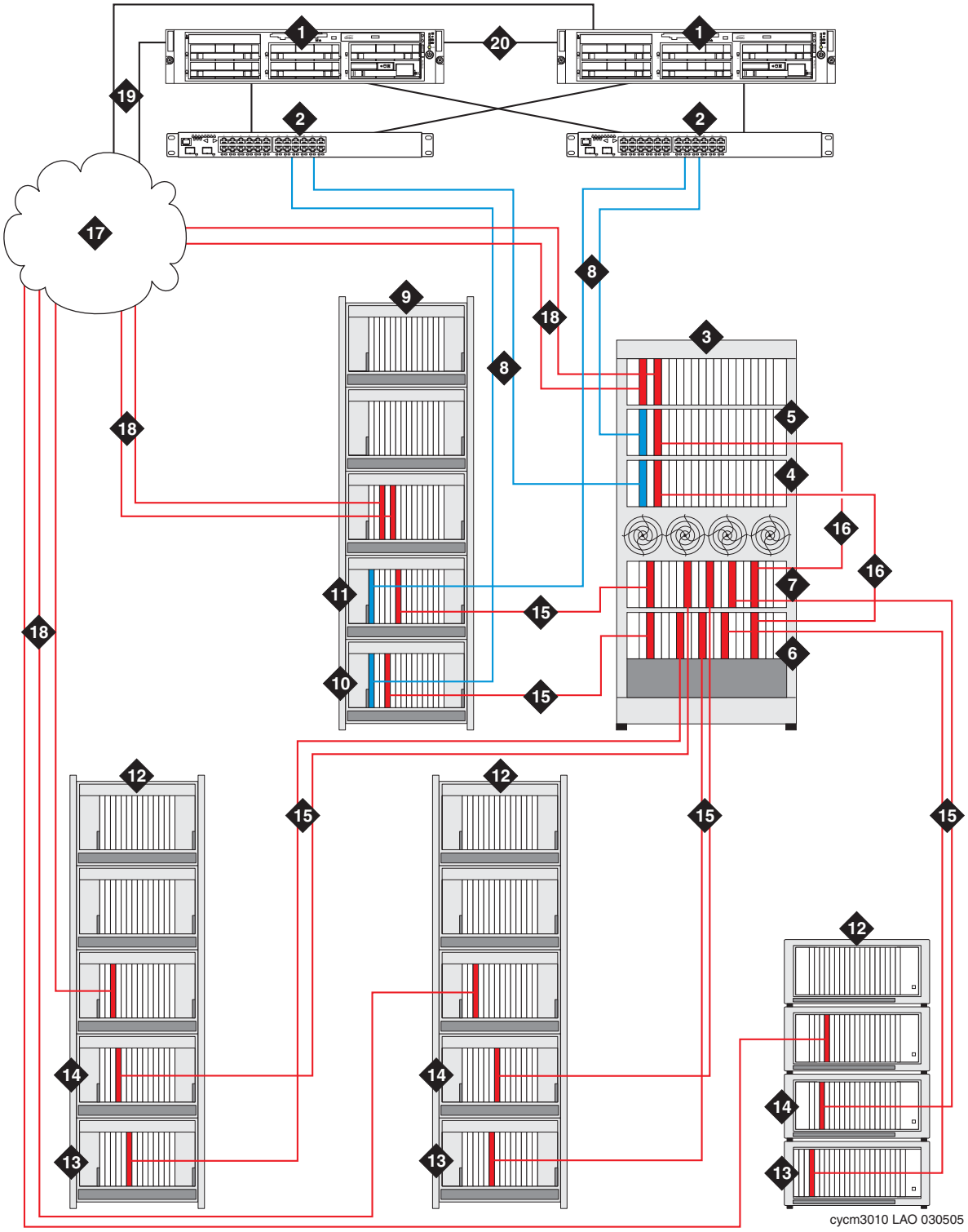

Figure notes: S8700-series Center Stage Switch duplicated control and duplicated voice-bearer networks

1. S8700-series Media Server
2. Ethernet Switch
3. IPSI-connected PN (MCC1 Media Gateway, consisting of at least two carriers).
4. Expansion port network (EPN) control carrier, in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.
   - A TN570B EI circuit pack for bearer and control network connections to the Switch Node Carrier (SNC).
5. Duplicated control carrier, in the B position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to duplicated control network.
   - A TN570B circuit pack for bearer and control network connections to the SNC.

     NOTE: For the duplicated control and bearer network configurations, each location of a PN or a group of PNs must contain a TN771 Maintenance Test circuit pack.
6. SNC, in the E position, which contains:
   - Multiple TN573B SNI circuit packs for EI connections to PNs
7. Duplicated SNC, in the D position, which duplicates the EI connections of the primary SNC.
8. Dedicated IPSI-to-server control network connection via Ethernet switch
9. IPSI-connected PN (G650 Media Gateway stack [shown in figure],MCC1 Media Gateway, or SCC1 Media Gateway stack), consisting of at least two media gateways or carriers).
10. PN control gateway or carrier, in the A position, which contains:
    - A TN2312AP/BP IPSI circuit pack for IP connection to server.

      NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.
    - A TN570B EI circuit pack for bearer and control network connections to the SNC.
11. Duplicated control gateway, in the B position, which contains:
    - A TN2312AP/BP IPSI circuit pack for IP connection to server.
    - A TN570B EI circuit pack for bearer and control network connections to the SNI.
12. Fiber-PNC PN (G650 Media Gateway stack [shown in figure], MCC1 Media Gateway, SCC1 Media Gateway stack [shown in figure]), consisting of at least two media gateways or carriers.
13. PN control gateway or carrier, in the A position, which contains:
    - A TN570B EI circuit pack for bearer and control network connections to the SNI.
    - One TN2182 Tone Clock circuit pack if the PN consists of SCC1 or MCC1 Media Gateways, or one maintenance-only TN2312AP/BP IPSI circuit pack if the PN(s) consist of G650 Media Gateways
14. Duplicated control gateway, in the B position, which contains:
    - A TN570B EI circuit pack for bearer and control network connections to the SNI.
    - One TN2182 Tone Clock circuit pack if the PN consists of SCC1 or MCC1 Media Gateways, or one maintenance-only TN2312AP/BP IPSI circuit pack if the PN(s) consist of G650 Media Gateways
15. TN 570B/573B fiber connections between PNs and SNCs
16. TN 573B/570B fiber connections between the SNCs and the MCC1's A and B carriers (if the MCC1 is a PN)
17. Customer LAN

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Figure notes: S8700-series Center Stage Switch duplicated control and duplicated voice-bearer networks (continued)

18. LAN connections, if any, of optional TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints

    NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, PNs, and adjunct systems. These circuit packs are optional for PNs in a CSS-connected network and may be inserted into a port carrier (shown in figure), the PN control carrier, or the duplicated control carrier. However, the C-LAN circuit pack is required for downloads of firmware updates.

19. LAN connections of media servers for remote administration

20. Duplicated server links, including the link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.

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S8700-series ATM Switch (single control network)

An S8700-series Media Server can support up to 64 PNs by using Asynchronous Transmission Mode (ATM) switching for PN connections. Each PN in the configuration must have a TN2305B ATM interface circuit pack (for multimode fiber) or a TN2306B ATM interface circuit pack (for single-mode fiber) in order to connect to every other PN in the system. The PNs can be MCC1, SCC1, or G650 Media Gateways (or gateway stacks). At least one PN is IPSI-connected to the S8700-series Media Servers. The ATM switch connects to the fiber with an OC-3 interface.

Note:

The ATM configuration illustrations show multi-mode fiber connections using TN2305B ATM-CES circuit packs and multi-mode fiber. With single-mode fiber connections, the configurations are the same, but the ATM connections uses TN2306B ATM-CES circuit packs and single-mode fiber.

A single IPSI circuit pack allows the server to control up to 5 PNs only. Therefore, in a configuration with many PNs, multiple PNs may be IPSI-connected.

TN2602AP circuit packs for duplicated bearer

For an S8700-series Media Server, any individual fiber-PNC PN can contain load-balancing or duplicated TN2602AP circuit packs. However, TN2602AP circuit packs do *not* need to be implemented uniformly within the system. Thus, some PNs may have no TN2602AP circuit pack, some PNs may have load-balancing TN2602AP circuit packs, and some PNs may have duplicated TN2602AP circuit packs. Thus, an S8700-series Media Server can have duplicated bearer connections, even though it does not support duplicated control or fiber-based duplicated bearer.

IPSI-connected PN

Only the G650 media gateway is available for new installations. However, because different migrations from older systems are supported, the PN connected to the S8700-series Media Server in an ATM configuration can consist of one of three gateways:

- G650 media gateway

  A G650 PN can consist of one to five G650 gateways in a stack connected by a TDM/LAN bus cable. One gateway, serving as control gateway in position A at the bottom of the stack, contains the following:

  - TN2312BP IPSI circuit pack
  - TN2305B or TN2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch
  - TN464GP DS-1 circuit pack for clock synchronization with a network resource

- SCC1 media gateway

  An SCC1 PN can consist of one to four SCC1 gateways in a stack connected by a TDM/LAN bus cable. One gateway, serving as control gateway in position A at the bottom of the stack, contains the following:

  - TN2312AP/BP IPSI circuit pack
  - TN2305B or TN2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch
  - TN464GP DS-1 circuit pack for clock synchronization with a network resource

  The control gateway or another gateway in the PN also contains a TN464GP DS-1 circuit pack for clock synchronization with a network resource

  Note:

  The TN2404 and TN2401 processor circuit packs, needed for the SI model, are *not* used and must be removed if the SCC1 is being migrated from a DEFINITY server.

- MCC1 media gateway

  An MCC1 PN has from one to five carriers in an MCC1 gateway connected by a TDM/LAN bus cables. One carrier, serving as control carrier in position A in the middle of the stack, contains the following:

  - TN2312AP/BP IPSI circuit pack
  - TN2305B or TN2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch

  The control carrier or another carrier in the PN also contains a TN464GP DS-1 circuit pack for clock synchronization with a network resource

Note:

The control carrier for a DEFINTY Server SI or R is not used and must be removed and replaced with an expansion control carrier if the MCC1 is being migrated. The processor circuit packs, needed for the SI or R models, are not used and must be removed. Other PNs can also be MCC1 Gateways.

PNs not IPSI-connected

In an ATM switch with a single control network configuration, additional PNs (up to 64) may be connected to the IPSI-connected PN using fiber optic cable between TN2305B/TN2306B ATM-CES circuit packs and an ATM switch. The cables are connected to the circuit packs using short-range or long-range multi-mode transceivers, or single-mode transceivers, depending on the distance between PNs.

The TN2305B/2306B ATM-CES circuit packs reside in the control carrier (MCC1) or control gateway (G650 or SCC1) of each PN. The control carrier or gateway in each additional PN also must contain a TN2182 Tone Clock circuit pack (SCC1 or MCC1) or a maintenance-only TN2312BP IPSI circuit pack (G650).

Figure 14: S8700-series ATM single control network

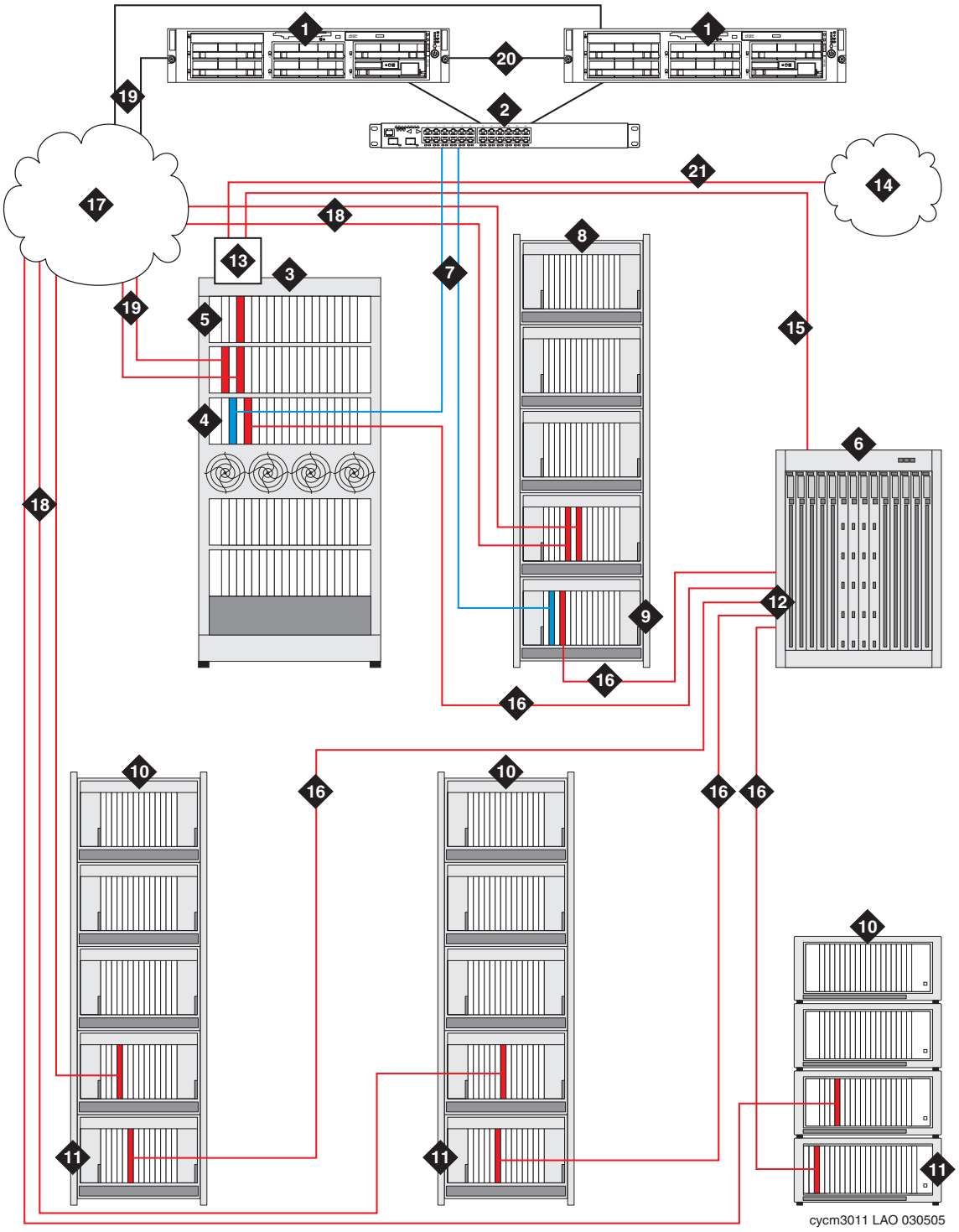

Figure notes: S8700-series ATM single control network

1. S8700-series Media Server
2. Ethernet Switch
3. IPSI-connect PN (G650 Media Gateway stack, MCC1 Media Gateway [shown in figure], or SCC1 Media Gateway stack), consisting of at least two media gateways or carriers.
4. PN control gateway or carrier, in the A position which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.

     NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.
   - A TN2305B or T2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch.
5. TN464GP DS-1 circuit pack, for clock synchronization with a network resource
6. ATM switch.
7. IPSI-to-server control network connection via Ethernet switch
8. IPSI-connect PN (G650 Media Gateway stack [shown in figure],MCC1 Media Gateway, or SCC1 Media Gateway stack).
9. PN control gateway or carrier, in the A position which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.

     NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.
   - A TN2305B or T2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch.
10. Fiber-PNC PN (G650 Media Gateway stack [shown in figure], MCC1 Media Gateway, SCC1 Media Gateway stack [shown in figure])
11. PN control gateway or carrier, in the A position, which contains:
    - A TN2305B or T2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch.

      NOTE: One TN2182 Tone Clock circuit pack must also be present per PN if the PN(s) consist of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312BP IPSI circuit pack must be present per PN if the PN(s) consist of G650 Media Gateways.
12. OC-3 connections to the ATM switch
13. 401A/B sync splitter, attached to the back of the TN464GP DS1 circuit pack
14. Public network (PSTN)
15. Timing signal to ATM switch from sync splitter.
16. Fiber connections from TN2305B/TN2306B to ATM switch.
17. Customer LAN
18. LAN connections, if any, of optional TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints. These circuit packs are optional for PNs in an ATM-connected network. However, the C-LAN circuit pack is required for downloads of firmware updates.
19. LAN connections of media servers for remote administration
20. Duplicated server links, including the link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.
21. DS1 connection from sync splitter.

S8700-series ATM Switch (duplicated control networks)

The high reliability ATM configuration duplicates the control network between the servers and the PNs. This configuration contains duplicated Ethernet switches, each connected to both S8700-series Media Servers. Remote IPSI-connected PNs also require duplicated Ethernet switches. However, IPSI-connected PNs that are collocated with the servers may share Ethernet switches with the server. The high reliability configuration also includes duplicated IPSIs in a second carrier or gateway of the IPSI-connected PN. In an ATM high reliability configuration, the *n + 1* formula for IPSIs is not required.

TN2602AP circuit packs for duplicated bearer

For an S8700-series Media Server, any individual fiber-PNC PN can contain load-balancing or duplicated TN2602AP circuit packs. However, TN2602AP circuit packs do *not* need to be implemented uniformly within the system. Thus, some PNs may have no TN2602AP circuit pack, some PNs may have load-balancing TN2602AP circuit packs, and some PNs may have duplicated TN2602AP circuit packs. Thus, an S8700-series Media Server can have duplicated bearer connections, even though it does not support duplicated control or fiber-based duplicated bearer.

Figure 15: S8700-series ATM duplicated control networks

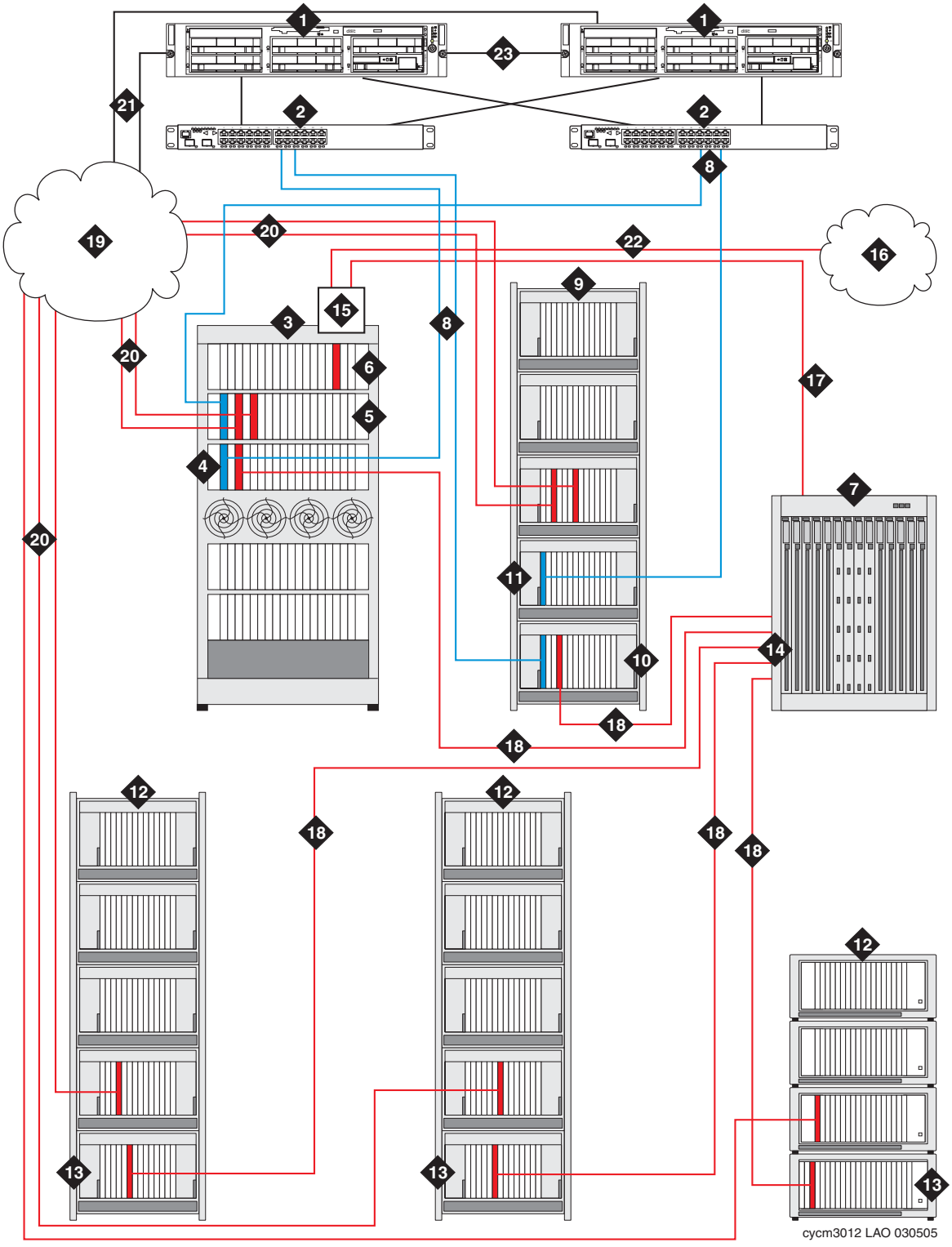

Figure notes: S8700-series ATM duplicated control networks

1. S8700-series Media Server
2. Ethernet Switch
3. IPSI-connected PN (G650 Media Gateway stack, MCC1 Media Gateway [shown in figure], or SCC1 Media Gateway stack), consisting of at least two media gateways or carriers.
4. PN control gateway or carrier, in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.

     NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.
   - A TN2305B or T2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch.
5. Duplicated control carrier, in the B position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to duplicated control network
6. TN464GP DS-1 circuit pack, for clock synchronization with a network resource
7. ATM switch.
8. IPSI-to-server control network connection via Ethernet switch
9. IPSI-connected PN (G650 Media Gateway stack [shown in figure],MCC1 Media Gateway, or SCC1 Media Gateway stack).
10. PN control gateway or carrier, in the A position, which contains:
    - A TN2312AP/BP IPSI circuit pack for IP connection to server.
    - A TN2305B or T2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch.
11. Duplicated control gateway, in the B position, which contains:
    - A TN2312AP/BP IPSI circuit pack for IP connection to server.
12. Fiber-PNC PN (G650 Media Gateway stack [shown in figure], MCC1 Media Gateway, SCC1 Media Gateway stack [shown in figure]).
13. PN control gateway or carrier, in the A position, which contains:
    - A TN2305B or T2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch.

      NOTE: One TN2182 Tone Clock circuit pack must also be present per PN if the PN(s) consist of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312BP IPSI circuit pack must be present per PN if the PN(s) consist of G650 Media Gateways.
14. OC-3 connections to the ATM switch
15. 401A/B sync splitter, attached to the back of the TN464GP DS1 circuit pack
16. Public network (PSTN)
17. Timing signal to ATM switch from sync splitter.
18. Fiber connections from TN2305B/TN2306B to ATM switch.
19. Customer LAN
20. LAN connections, if any, of optional TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints. These circuit packs are optional for PNs in an ATM-connected network. However, the C-LAN circuit pack is required for downloads of firmware updates.
21. LAN connections of media servers for remote administration
22. DS1 connection from sync splitter.
23. Duplicated server links, including the link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.

S8700-series ATM Switch (duplicated control and bearer networks)

Like the high reliability ATM configuration, the critical reliability ATM configuration duplicates the control network between the servers and the PNs. In addition, the critical reliability configuration contains duplicated ATM switches and ATM connections, with each PN containing duplicated TN2305B/TN2306B ATM-CES circuit packs with connections to both ATM switches. In addition, each non-IPSI-connected PN must have duplicate TN2182CTone Clock circuit packs. And finally, in each location of a PN or group of PNs, one of the PNs must have a TN771 Maintenance Test circuit pack.

As with an ATM high reliability configuration, the *n + 1* formula for IPSIs is not required.

TN2602AP circuit packs for duplicated bearer

For an S8700-series Media Server, any individual fiber-PNC PN can contain load-balancing or duplicated TN2602AP circuit packs. However, TN2602AP circuit packs do *not* need to be implemented uniformly within the system. Thus, some PNs may have no TN2602AP circuit pack, some PNs may have load-balancing TN2602AP circuit packs, and some PNs may have duplicated TN2602AP circuit packs. Thus, an S8700-series Media Server can have duplicated bearer connections, even though it does not support duplicated control or fiber-based duplicated bearer.

Figure 16: S8700-series ATM duplicated control and duplicated voice-bearer networks

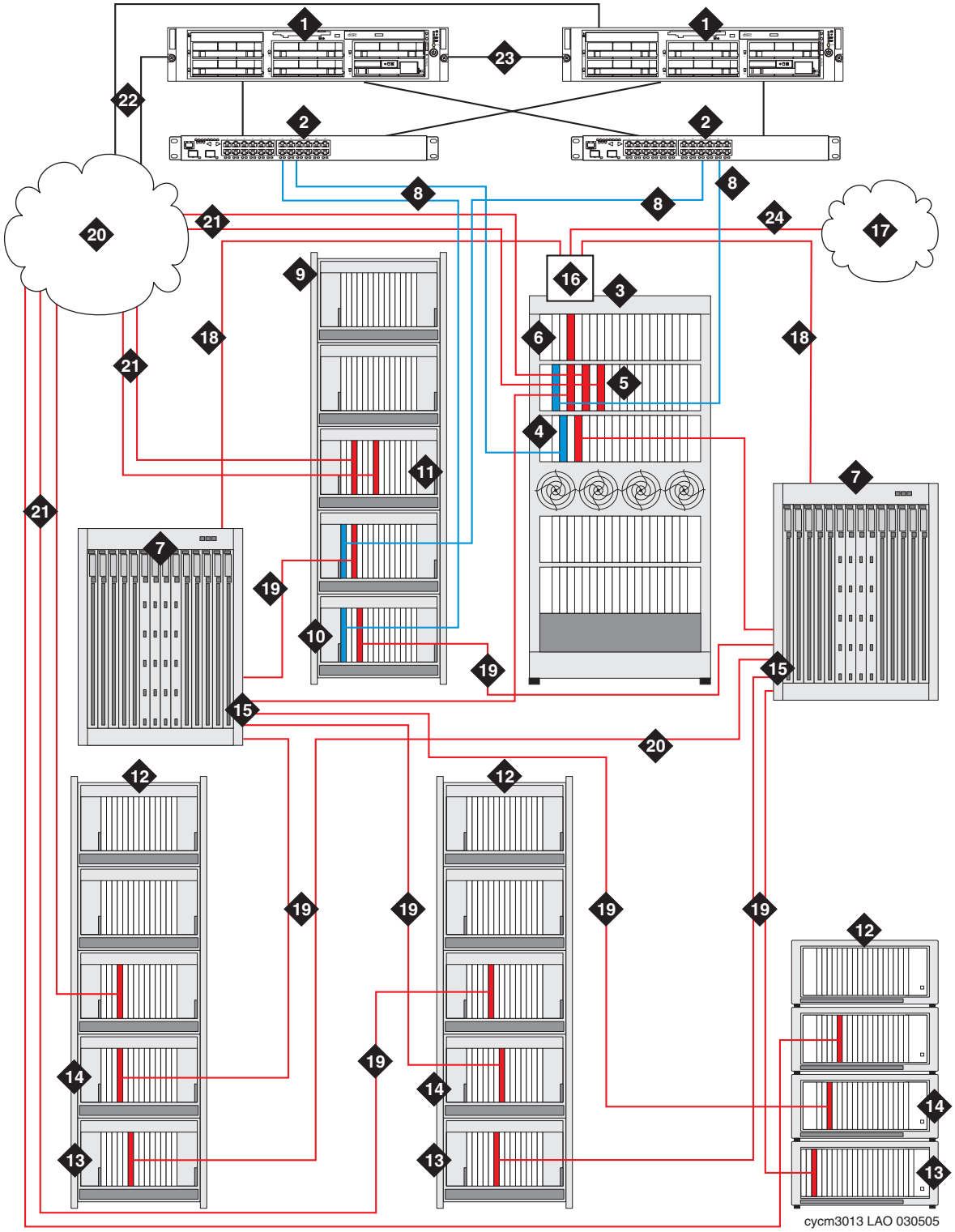

Figure notes: S8700-series ATM duplicated control and duplicated voice-bearer networks

1. S8700-series Media Server
2. Ethernet Switch
3. IPSI-connect PN (G650 Media Gateway stack, MCC1 Media Gateway [shown in figure], or SCC1 Media Gateway stack), consisting of at least two media gateways or carriers.
4. PN control gateway or carrier, in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.

     NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.

   - A TN2305B or T2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch.
5. Duplicated control carrier or gateway, in the B position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to duplicated control network
   - A TN2305B (for multimode fiber) or TN2306B (for single-mode fiber) ATM-CES circuit pack for bearer and control network connections to the duplicated ATM switch.

     NOTE: For the duplicated control and bearer network configurations, each location of a PN or a group of PNs must contain a TN771 Maintenance Test circuit pack.
6. TN464GP DS-1 circuit pack, for clock synchronization with a network resource
7. ATM switch. There are two ATM switches in this configuration.
8. IPSI-to-server control network connection via Ethernet switch
9. IPSI-connected PN (G650 Media Gateway stack [shown in figure],MCC1 Media Gateway, or SCC1 Media Gateway stack, consisting of at least two media gateways or carriers).
10. PN control gateway or carrier, in the A position, which contains:
    - A TN2312AP/BP IPSI circuit pack for IP connection to server.
    - A TN2305B or T2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch.
11. Duplicated control gateway or carrier, in the B position, which contains:
    - A TN2312AP/BP IPSI circuit pack for IP connection to server.
    - A TN2305B or TN2306B ATM-CES circuit pack for bearer and control network connections to the duplicated ATM switch.
12. Fiber-PNC PN (G650 Media Gateway stack [shown in figure], MCC1 Media Gateway, SCC1 Media Gateway stack [shown in figure]), consisting of at least two media gateways or carriers.
13. PN control gateway or carrier, in the A position which contains:
    - A TN2305B or T2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch.
    - One TN2182 Tone Clock circuit pack if the PN consists of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312BP IPSI circuit pack if the PN consists of G650 Media Gateways.
14. Duplicated control gateway or carrier, in the B position which contains:
    - A TN2305B or T2306B ATM-CES circuit pack for bearer and control network connections to the ATM switch.
    - One TN2182 Tone Clock circuit pack if the PN consists of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312BP IPSI circuit pack if the PN consists of G650 Media Gateways.
15. OC-3 connections to the ATM switch
16. 401A/B sync splitter, attached to the back of the TN464GP DS1 circuit pack
17. Public network (PSTN)

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Figure notes: S8700-series ATM duplicated control and duplicated voice-bearer networks (continued)

18. Timing signal to ATM switch from sync splitter.
19. Fiber connections from TN2305B/TN2306B to ATM switch.
20. Customer LAN
21. LAN connections, if any, of optional TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints. These circuit packs are optional for PNs in an ATM-connected network. However, the C-LAN circuit pack is required for downloads of firmware updates.
22. LAN connections of media servers for remote administration
23. Duplicated server links, including the link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.
24. DS1 connection from sync splitter.

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Distance options with fiber-optic connections

Fiber connections up to 200 feet (61 meters)

EI-to-EI or EI-to-SNI intercabinet connections are implemented by installing a lightwave transceiver on the I/O connector plate for each of the administered fiber endpoints. Each lightwave transceiver has a receive and a transmit connector for either a 62.5-micron or 50-micron fiber connection. Standard fibers are available in various lengths up to 150 feet (46 m) for single-mode fiber and up to 200 feet (61 m) for multimode fiber. These fibers are used to connect lightwave transceivers to each other when they are close enough together, or to optical cross-connect facilities for greater distances.

See TN570B Expansion Interface PN connections up to 200 ft. on page 75.

Figure 17: TN570B Expansion Interface PN connections up to 200 ft.

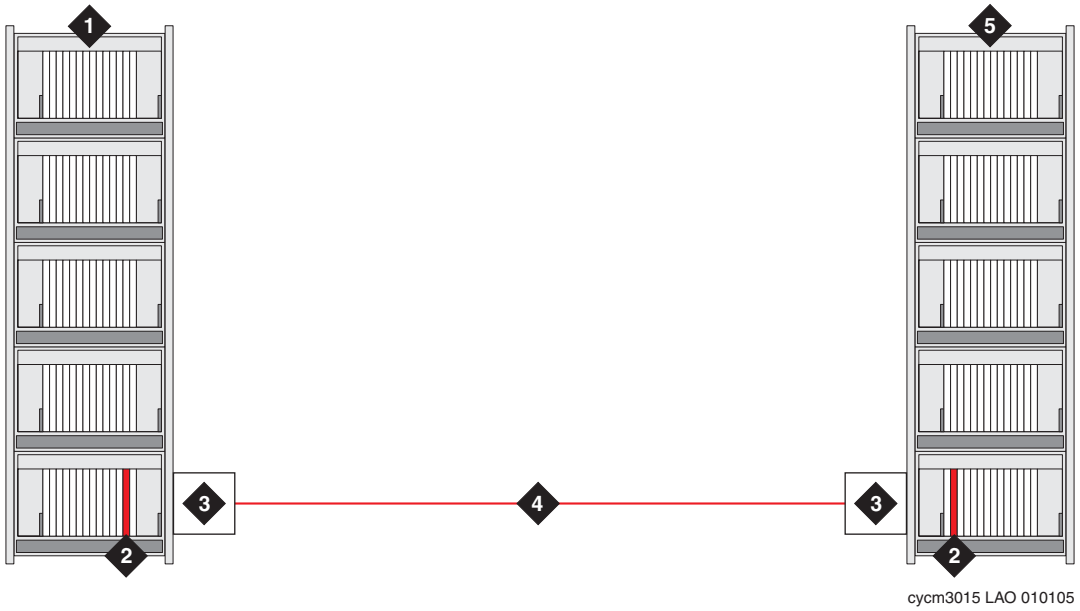

Figure notes:

1. Local PN
2. TN570B Expansion Interface circuit pack
3. 9823A short range transceiver
4. Multimode fiber cable

Fiber connections up to 22 miles (35.4 kilometers)

The lightwave transceivers are powered from I/O connector plate leads attached to TN570 Expansion Interface circuit pack or a TN573 SNI circuit pack. The transceivers include loop-around capabilities to support fiber fault isolation. Table 2 lists part number and distance specifications for the two length-dependent 9823-type multimode transceivers and the 300A single mode fiber transceiver. The transceivers at each end of the fiber should match.

Table 2: Lightwave transceiver specifications

Lightwave transceiver part number	Maximum fiber length	Fiber mode
9823A	4900 feet (1494 m)	Multimode
9823B	25,000 feet (7620 m)	Multimode
300A	22 miles (35.4 km)	Single mode

Figure 18: TN570B Expansion Interface PN connections up to 4900/25000 ft. and 22 miles.

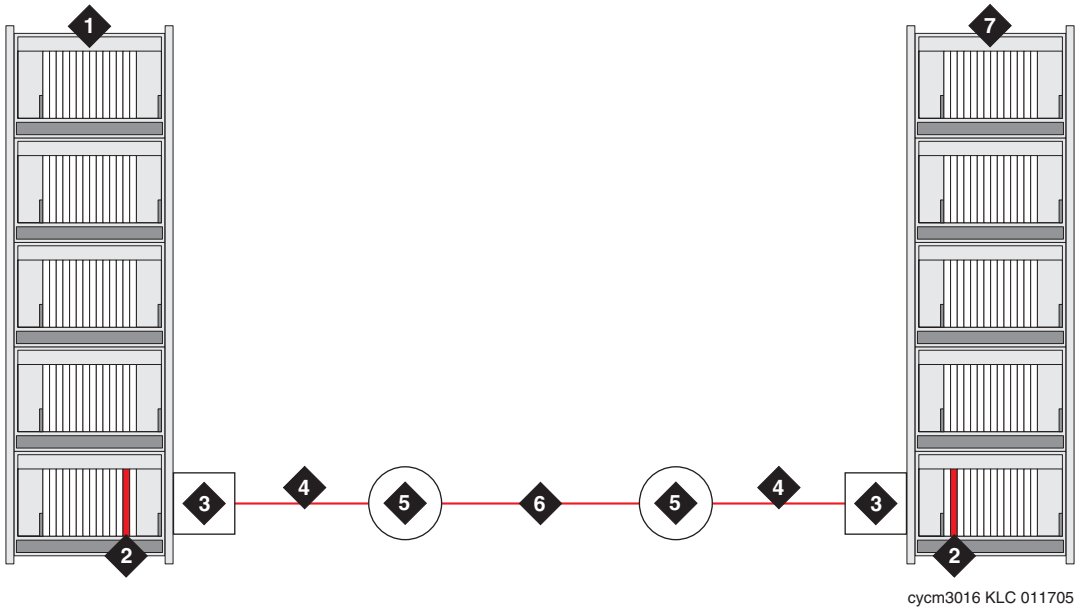

Figure notes:

1. Local PN
2. TN570B Expansion Interface circuit pack
3. 9823A short range transceiver (up to 4900 ft. or 1494 m), 9823B long range transceiver (up to 25000 ft. or 7620 m), or 300A transceiver (22 miles or 35.4 km)
4. Optic fiber
5. Lightguide Interconnect Unit (LIU)
6. Single-mode or multimode fiber cable

Fiber connection up to 200 miles

When fiber-optic cabling is not practical, Digital Service 1 (DS1) can be used to connect PNs up to 200 miles (322 km) apart. A TN574 or TN1654 DS1 Converter (DS1 CONV) circuit pack serves as the interface between the network and an EI or SNI on the switch. DS1 cabling on a carrier consists of a Y-cable that connects a DS1 CONV to an EI or SNI and to the network.

Table 3 lists the lengths and uses for DS1 CONV cables, depending upon where the DS1 CONV and the EI or SNI are located.

Table 3: DS1 CONV cable specifications

Connection location	Length
On same half carrier	1 foot (30.48 cm)
On different half carriers in same cabinet	5.5 feet (1.68 m)
Between two adjacent cabinets	1 foot (30.48 cm), used with two 9823As, and 1 20-foot (6.1 m) fiber-optic cable

The DS1 CONV to EI/SNI cable is a shielded metallic Y-cable held in place at the EI/SNI port connector by a 4B retainer and at the DS1 CONV port connector by a 4C retainer. The cable end with one 25-pair amphenol connector attaches to the I/O Plate connector for the EI or SNI. The end with two 25-pair amphenol connectors attaches to the DS1 CONV I/O plate connector.

The 13-inch (33-centimeter) cable 846448652 or 847245776 connects the DS1 CONV to a fiber-optic cable, enabling the DS1 CONV to connect to an EI or SNI at a greater distance. The cable end with one 25-pair amphenol connector attaches to a lightwave transceiver using the 846885259 bracket. The end with two 25-pair amphenol connectors attaches to the DS1 CONV I/O plate connector. The other end of the fiber-optic cable connects to a lightwave transceiver attached to the I/O plate connector of the EI or SNI.

An H600-348 cable connects the DS1 CONV cable to a CSU (channel service unit), which connects to a wall field. Alternatively, connection is sometimes made directly from the Y-cable to the wall field. This cable provides from one to four DS1 connections. One end of the H600-348 cable is plugged into the 50-pin amphenol piggy-back connector on the 8464486xx cable connected to the DS1 CONV port connector. The other end of the H600-348 cable has four 15-pin sub-miniature D-type connectors that plug into the CSU. Table 4 lists the H600-348 cable specifications.

Table 4: H600-348 cable specifications

Group No.	Length	Group No.	Length
G1	25 feet (7.62 m)	G5	125 feet (38.1 m)
G2	50 feet (15.24 m)	G6	200 feet (60.96 m)
G3	75 feet (22.86 m)	G7	400 feet (121.9 m)
G4	100 feet (30.48 m)	G8	650 feet (198 m)

See TN1654 DS1 Converter/TN570B Expansion Interface PN connections up to 200 miles. on page 78.

Figure 19: TN1654 DS1 Converter/TN570B Expansion Interface PN connections up to 200 miles.

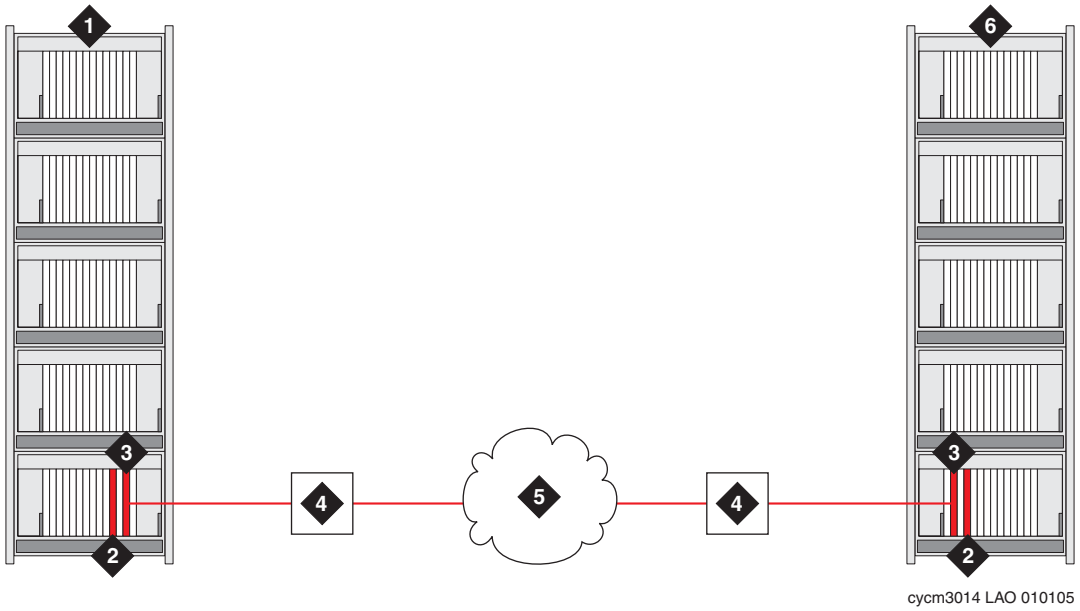

Figure notes:

1. Local PN
2. TN570B Expansion Interface circuit pack
3. TN1654 DS1 Converter circuit pack
4. Channel service units (up to 4), each with at T1 trunk
5. Public network (PSTN)

Metallic cable for intracabinet connections

Metallic cable can be substituted for fiber-optic cable for “fiber” connections between EIs or between an EI and an SNI in the same MCC cabinet, using the same I/O plate connectors.

DANGER:
The metallic cables should not be used for intercabinet connections, since doing so would violate system ground integrity.

Table 5 lists the part numbers and uses for the two (2) metallic cable lengths.

Table 5: Metallic cable specifications

Metallic cable part numbers	Length	Use
H600-278,G1	13 inches (33 cm)	From an EI in slot 1 of a switch node carrier to an SNI in the same half of the carrier (usually the adjacent slot)
H600-278,G2	66 inches (168 cm)	From an EI to an SNI in the same cabinet, but in a different carrier or different half of a carrier

Configurations with both IP-PNC and fiber-PNC PNs

Communication Manager R3.1 allows the S8700-series, S8500, S8500B, and S8500C Media Servers to support configurations that combine IP-PNC port networks (PNs) with direct-connect PNs, CSS-connected PNs or ATM-connected PNs. Additionally, Communication Manager R3.1 allows the media servers to support configurations that contain both single control networks and duplicated control networks and both single bearer networks and duplicated bearer networks.

This capability allows customers to do the following:

- Add IP PNs to a fiber-PNC configuration using the simpler, less costly connections over the customer LAN. In this way, customers can avoid the complication and cost of adding fiber-PNC PNs. This capability can be especially attractive when it eliminates the need for installing a DS1C circuit pack and a connection over a T1 trunk to the new PN.
- Convert and consolidate, in an easy, cost-effective way, remote standalone DEFINITY servers (R, SI, CSI, or S8100) and their PNs into a single network of PNs controlled by, and administered with, one server.
- Configure, within the single footprint of an MCC1 Media Gateway, multiple port networks, using IP-PNC, fiber-PNC PNs, or a variety of combinations of the two. In this way, customers have tremendous flexibility in configuring MCC1 Media Gateways to balance reliability, call capacities and feature richness.
- Configure reliability into a network in a more cost-effective, flexible way. Duplication of control and bearer networks can be configured based on the criticality of the location or the needs of users connected to a particular PN.

  Note:

  All port networks that are fiber-PNC within a direct-connect, CSS or ATM switch configuration must still have the same reliability level — all single control and bearer network, all duplicated control network, or all duplicated control and bearer network.

Possibilities for combining IP-PNC and fiber-PNC PNs in a configuration

A Communication Manager configuration can contain one of the following combinations of port network connection methods:

- IP-PNC and direct-connect — available with S8500, S8500B, S8500C, S8710 or S8720 Media Servers as main servers, but not as Enterprise Survivable Servers (ESSs)
- IP-PNC and Center Stage Switch (CSS) — available with S8700-series Media Servers as both main servers and ESSs
- IP-PNC and Asynchronous Transmission Mode (ATM) — available with S8700-series Media Servers as both main servers and ESSs

Note:

You *cannot* mix CSS and ATM port network connections in the same configuration. You also *cannot* mix direct-connect PN connections with ATM or CSS port network configurations.

Note:

The DEFINITY Server CSI does not support multiple port networks and, therefore, does not support combining PN connection methods.

Regardless of the combinations of PN connection methods, the maximum number of PNs allowed continues to be 64. However, since a server can support IP-PNC and CSS PNs simultaneously, the following capacity rules apply to a configuration with both IP-PNC and fiber-PNC PNs:

- With CSS, two to 44 CSS PNs, with additional IP-PNC PNs for a maximum total of 64 PNs
- With ATM, 64 ATM and IP-PNC PNs in any combination
- With direct-connect, two to three direct-connect PNs, with 1 to 62 IP-PNC PNs, for a maximum total of 64 PNs

Media gateway combinations

Like the G650 Media Gateway, SCC1 and MCC1 Media Gateways can connect to other port networks using either IP-PNC or fiber-PNC options (direct/CSS/ATM-connect). The CMC1 and G600 media gateways can be IP-PNC only and cannot be fiber-PNC in any configuration. But, because a server can use the IP-PNC method with any of the direct, CSS, or ATM connection methods simultaneously, the server can simultaneously connect CMC1s and/or G600s as IP-PNC media gateways in the same network that includes direct, CSS, or ATM-connected G650, SCC1, and/or MCC1 Media Gateways. As a result, a configuration with IP-PNC and fiber-PNC PNs may contain any or all media gateways that are supported by the current release of Communication Manager. The following table lists, by server, the media gateways and

connection methods that the servers can simultaneously support in a port network configuration.

Server	Supported Media Gateways	IP-PNC	Direct-connect	CSS/ ATM-connect [1]	Reliabilities supported
S8500C/ S8500B	CMC1	yes	no	no	single control and bearer only
	G600	yes	no	no	same as CMC1
	G650	yes	yes	no	single control and bearer, single control and duplicated bearer
	SCC1	yes	yes	no	same as G650
	MCC1	yes	yes	no	same as G650
S8710/S8720	CMC1	yes	no	no	single control and bearer only
	G600	yes	no	no	same as CMC1
	G650	yes	yes	yes (requires an MCC1 for SNC/CSS)	single control and bearer, duplicated control only, single control and duplicated bearer, and duplicated control and bearer
	SCC1	yes	yes	yes (requires an MCC1 for SNC/CSS)	same as G650
	MCC1	yes	yes	yes	same as G650

1. For any system, either CSS or ATM connections may be used, but not both.

Options for multiple levels of reliability

Within the fiber-PNC portion of a system (direct-, CSS, or ATM-connected PNs), duplicated bearer reliability using fiber must be uniformly applied, and IPSI duplication must also be uniform among the PNs that have IPSIs. However, a mixed configuration of IP-PNC and fiber-PNC PNs may collectively have multiple levels of reliability. The system-wide network of PNs does not have to be "all duplicated IPSI" or "all simplex IPSI," or "all duplicated bearer" or "all simplex bearer." TN2602AP circuit pack duplication does not have to be uniform.

Administering an S8700-series Media Server for duplicated and single control networks

With direct/CSS/ATM PN connections and duplicated control networks, an S8700-series Media Server’s control network A and control network B interfaces are administered as dedicated control networks and connected to duplicated IPSI circuit packs in the fiber-PNC PNs. If a remote IP-PNC PN is introduced into the configuration, the S8700-series Media Server and IP-PNC PN is administered for a control network over the customer’s LAN. In this case, a third control network C may be administered on the S8700-series Media Server. The S8700-series Media Server automatically uses its own customer LAN interface port for Control network C.Therefore, to administer control network C for IP-PNC PNs, you only have to tell the media server to turn on control network C.

Dedicated and non-dedicated control networks

Control networks A and B can be separately configured for dedicated and non-dedicated control networks. You can also use control network C to connect IP-PNC PNs, while using control networks A and B for dedicated control networks with the fiber-PNC PNs. Control network C uses the customer LAN exclusively for the control signaling, while control networks A and B may use either dedicated Ethernet switch connections or the customer LAN for control signaling.

Requirements for using both IP-PNC and fiber-PNC PNs

A configuration that has both IP-PNC and fiber-PNC PNs requires the following:

- A Communication Manager license that has IP port network Connectivity (IP-PNC) turned off (that is the feature keyword in the license file, FEAT_IP_PNC, is off and the **IP PNC?** field on the Customer Options screen is **n**)

  Communication Manager allows IP-PNC PNs to be added to an existing fiber-PNC configuration because IP-PNC is already turned off.

CAUTION:

If you want to convert or migrate fiber-PNC SCC1 or MCC1 PNs to IP-PNC PNs, the RFA license file entry, FEAT_IP_PNC keyword, must be ***off***.

- At least one TN2302AP IP Media Processor or TN2602AP IP Media Resource 320 circuit pack in a PN in the fiber-PNC PN configuration.

  Since an IP-PNC PN does not have fiber connections with the direct, CSS, or ATM PNs, bearer transmission between IP-PNC PNs and direct/CSS/ATM PNs must occur over the IP network. Because they convert TDM calls to IP, and IP calls to TDM, the TN2302AP or TN2602AP circuit packs enable bearer transmission over IP networks.

- The fiber-PNC PN or PNs that contain the TN2302AP or TN2602AP circuit packs serve as gateways between the IP-PNC and fiber-PNC portions of the configuration. As a result, the gateway TN2602AP circuit pack or circuit packs must be in a network region that can reach, and is reachable by, TN2602AP circuit packs in any and all IP-PNC PNs. To be reachable, the gateway TN2602AP circuit packs can be one or both of the following:

  - In the same network region as the TN2602AP or TN2302BP circuit packs of other PNs.
  - Mapped to the IP PNC addresses in other network regions.

- Like IP-PNC PNs, fiber-PNC PNs can have up to two TN2602AP circuit packs installed. In addition, in a fiber-PNC PN, the TN2602AP circuit packs can be either in load-balancing mode or in duplicated bearer mode.

CAUTION:

The addition of a TN2302AP or TN2602AP circuit pack to a fiber-PNC PN may have a significant impact on traffic that must be handled by the PN. That is, in some scenarios, the PN may not have enough timeslot availability.

For example, the targets of a large number of IP station or trunk calls may be TDM stations or trunks in fiber-PNC PNs that do not have TN2302AP/TN2602AP circuit packs. In this case, the talk paths are routed through a fiber-PNC PN containing the TN2302AP/TN2602AP circuit packs. This routing may exhaust the 484 time slots of the gateway PN and cause calls to be blocked.

To avoid this media processor linking, TN2302AP/TN2602AP circuit packs should generally be placed in every fiber-PNC PN. This need becomes more apparent when the gateway PN uses a 320-channel TN2602 instead of an 80-channel TN2602.

As a result, you should analyze the traffic measurements on such PNs prior to configuring a PN as a IP-to-TDM gateway.

Note:

IP-PNC PNs always require at least one TN2302AP or TN2602AP circuit pack.

TN2602AP circuit packs in fiber-PNC PNs

Any fiber-PNC PN can optionally contain TN2302BP or TN2602AP circuit packs in order to support IP endpoints and trunks. However, to combine IP-PNC PNs in a configuration with fiber-PNC PNs, a TN2602AP (or optionally, a TN2302BP) circuit pack is *required* in at least one of the fiber-PNC PNs that also contains an IPSI connection. A fiber-PNC PN with one of these circuit packs can then serve as a gateway between the fiber-PNC and IP-PNC portions of the Communication Manager configuration so the portions can communicate via the LAN/WAN.

Any individual fiber-PNC PN can also contain load-balancing or duplicated TN2602AP circuit packs. TN2602AP circuit packs do *not* need to be implemented uniformly within the fiber-PNC portion of the system. Thus, some PNs may have no TN2602AP circuit packs, some PNs may have load-balancing TN2602AP circuit packs, and some PNs may have duplicated TN2602AP circuit packs. Duplicated TN2602AP circuit packs can provide duplicated bearer capability when the fiber connections do not.

Examples of combining IP-PNC and fiber-PNC PNs

The following sample configurations illustrate some examples of combining IP-PNC and fiber-PNC PNs. Some examples also illustrate combining different reliability levels.

Example of combining direct- and IP-PNC PNs

Figure 20 illustrates an S8500 Media Server configuration that combines direct-connect PNs with IP-PNC PNs. The IP-PNC PN is labeled as item 11. The other PNs, items 3 and 5, are direct-connect PNs.

Figure 20: Direct- and IP-PNC PNs example (with S8500 Media Server)

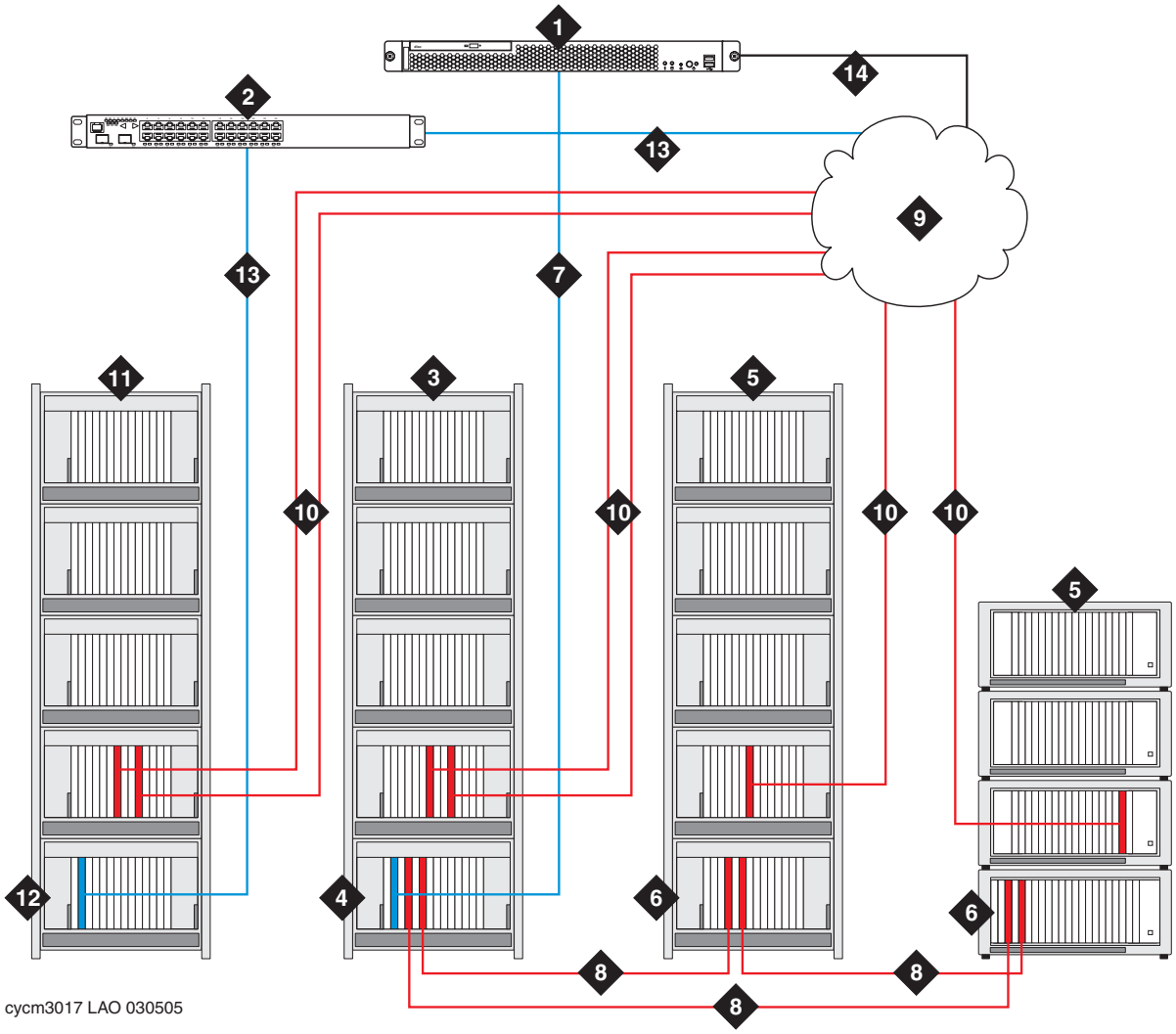

Figure notes: Direct- and IP-PNC PNs example (with S8500 Media Server)

1. S8500C or S8500B Media Server
2. LAN connections of media server for remote administration
3. IPSI-connected port network (G650 Media Gateway or G650 stack [shown in figure], MCC1 Media Gateway or SCC1 Media Gateway or SCC1 stack). The PN is part of the fiber-PNC bearer network.

   NOTE: G600 or CMC1 Media Gateways can be used in IP-PNC configurations only.
4. PN control gateway or carrier, in the A position in PN 3, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.
   - Two TN570Bv7/C/D EI circuit packs for bearer network connections to the other two PNs (if any).
5. Fiber-PNC PN (G650 Media Gateway or G650 stack [shown in figure], MCC1 Media Gateway, or SCC1 Media Gateway or SCC1 stack [shown in figure]).

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Figure notes: Direct- and IP-PNC PNs example (with S8500 Media Server) (continued)

6. PN control gateway or carrier within PNs labeled 5, in the A position in the gateway stack or MCC1. The control gateway contains two TN570Bv7/C/D EI circuit packs for bearer network connections to the other two PNs.

   NOTE: One TN2182 Tone Clock circuit pack must also be present per PN if the PN(s) consist of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312AP/BP IPSI circuit pack must be present per PN if the PN(s) consist of G650 Media Gateways.

7. IPSI-to-server control network connection. Requires dual NIC card on the media server.

8. TN 570Bv7/C/D to 570Bv7/C/D fiber connections between PNs

9. Customer LAN

10. LAN connections, if any, of TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints and firmware downloads

    NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, port networks, and adjunct systems. These circuit packs may be inserted into a port carrier (shown in figure), the PN control carrier, or the duplicated control carrier.

11. IP-PNC PN (G650 Media Gateway or stack [shown in figure]). May also be a G600, SCC1, MCC1, or CMC1 from an S8100 or DEFINITY Server migration.

12. Control gateway in PN 11, in the A position in the gateway stack. The control gateway contains:
    - A TN2312AP/BP IPSI circuit pack for IP connection to server.

13. IPSI-to-server control network connection through Ethernet switch and customer LAN.

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Example of IP-PNC PNs with different reliability levels

Figure 21 illustrates an S8700-series Media Server configuration that combines duplicated control/duplicated bearer network, duplicated control-only network, and single control network reliability configurations in an IP-PNC network. The PN with a single control network is labeled as item 11. Other PNs, items 3, have duplicated control networks.

Figure 21: IP-PNC PNs with single control network, duplicated control networks, and duplicated control/bearer network example (with S8700-series Media Server)

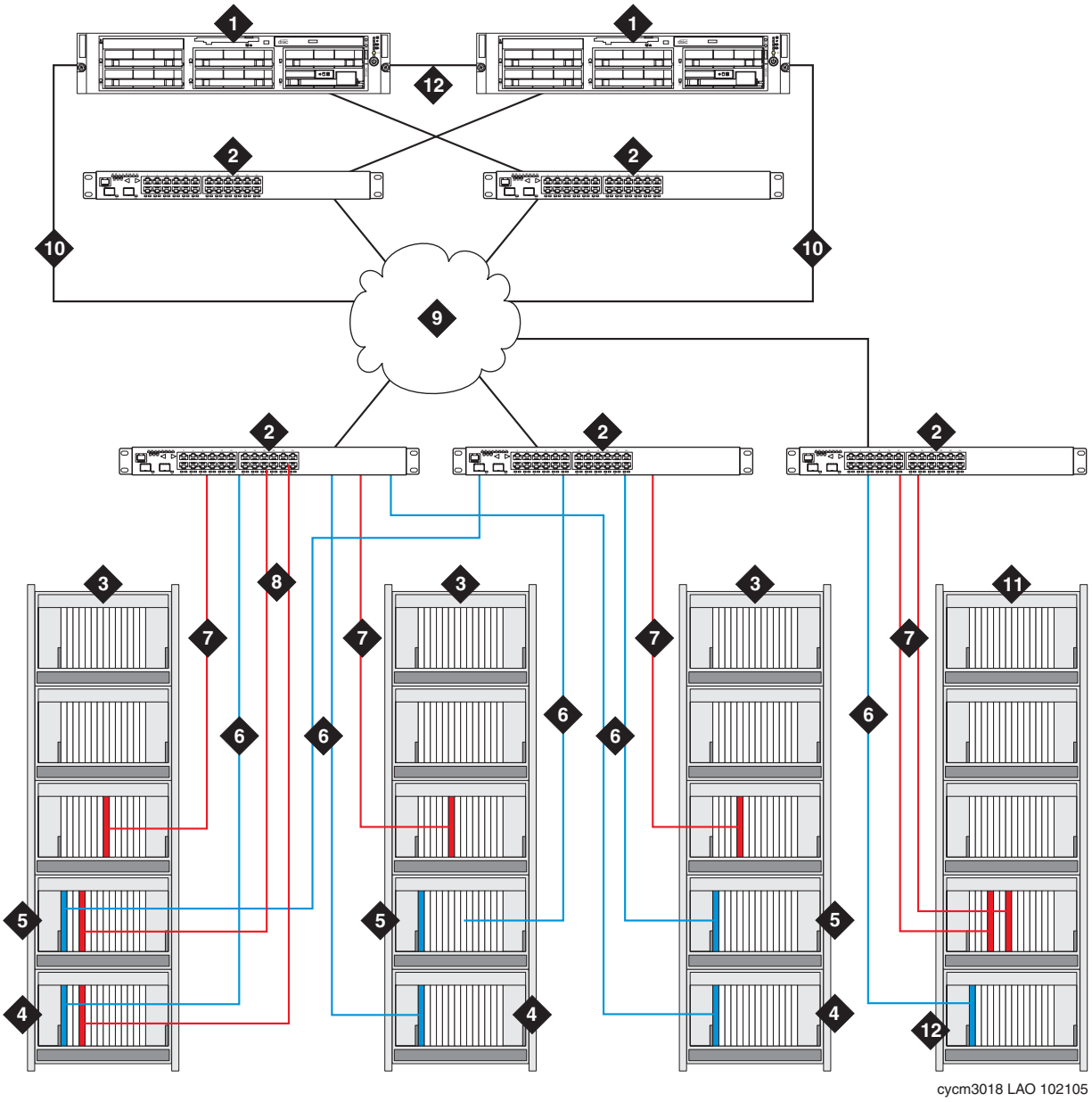

Figure notes: IP-PNC PNs with single, duplicated control networks, and duplicated control/bearer network (with S8700-series Media Server)

1. S8700-series Media Server
2. Ethernet Switch. For local LAN connections, the same pair of Ethernet switches may connect both the media servers and the media gateways. For remote LAN/WAN connections, the remote gateway(s) must have a pair of Ethernet switches at the remote location.

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Figure notes: IP-PNC PNs with single, duplicated control networks, and duplicated control/bearer network (with S8700-series Media Server) (continued)

3. IP-PNC PNs (G650 Media Gateway or stack [shown in figure]). May also be SCC1 or MCC1 Media Gateways from a DEFINITY Server migration.
4. Control gateway for PN 3, in the A position in the gateway stack. The control gateway contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.
5. Duplicated PN control gateway for PN3, in the B position in the gateway stack. The control gateway contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to control network.
6. IPSI-to-server control network connection via Ethernet switch
7. LAN connections of TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints

   NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, port networks, and adjunct systems. These circuit packs may be inserted into a port carrier (shown in figure), the PN control carrier, or the duplicated control carrier.
8. Customer LAN
9. LAN connections of media servers for remote administration
10. Duplicated server links, including the fiber link for translations transfer and the DAL2 link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication
11. IP-PNC PN (G650 Media Gateway or stack [shown in figure]). May also be a G600 Media Gateway or stack or a CMC1 from an S8100 Media Server or a DEFINITY Server migration, an MCC1 Media Gateway from a DEFINITY Server migration, or an SCC1 Media Gateway.
12. PN control gateway, in the A position in the gateway stack, for PN 11. The control gateway contains:
    - A TN2312AP/BP IPSI circuit pack for IP connection to server.

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Example of combining IP- and fiber-PNC PNs with different reliability levels

Figure 22 illustrates an S8700-series Media Server configuration that combines the following:

- Fiber-PNC PNs (CSS-connected PNs in this example) with standard single control network reliability and duplicated (item 3 in Figure 22), load-balancing (item 7 in Figure 22), single (item 9 in Figure 22), and no (item 21 in Figure 22) TN2602AP circuit packs.
- An IP-PNC PN (item 17 in Figure 22) with duplicated control and duplicated bearer network reliability.

Note:

The IP-PNC PN (item 17 in Figure 22) is connected to two Ethernet switches on the customer network since the CSS-connected PNs are connected to the server over a single dedicated Ethernet switch.

Figure 22: CSS-connected PNs (single control network) and IP-PNC PNs (duplicated control and duplicated bearer network) example

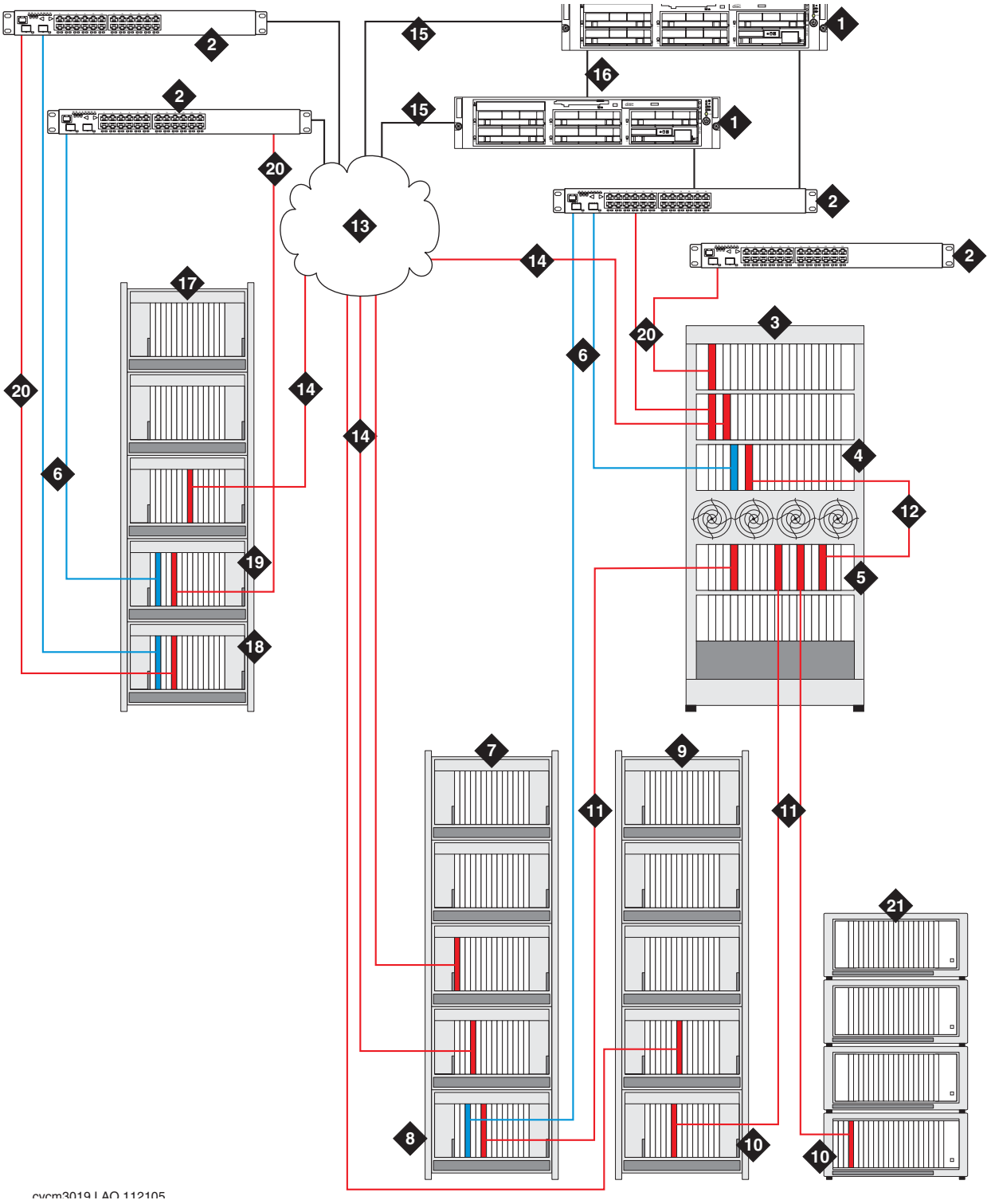

Figure notes: CSS-connected PNs (single control network) and IP-PNC PNs (duplicated control and duplicated bearer network) example

1. S8700/S8710/S8720 Media Server
2. Ethernet Switch
3. Fiber-PNC MCC1 Media Gateway (CSS and PN) with duplicated TN2602AP circuit packs. This PN serves as the gateway PN to IP-PNC PNs.
4. Control carrier for PN 3, in the A position in the MCC1. The control carrier contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.
   - A TN570Bv7/C/D EI circuit pack for bearer network connections to the Switch Node Carrier (SNC).
5. Switch node carrier (SNC), which contains:
   - Multiple TN573 SNI circuit packs for EI connections to PNs
6. IPSI-to-server control network connection via Ethernet switch
7. Second fiber-PNC and IPSI-connected PN (G650 Media Gateway or stack [shown in figure], MCC1 Media Gateway, or SCC1 Media Gateway stack). This PN has load-balancing TN2602AP circuit packs and also serves as a gateway to IP-PNC PNs.
8. Control gateway or carrier for PN 7, in the A position in the stack. The control gateway or carrier contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.
   - A TN570Bv7/C/D EI circuit pack for bearer network connections to the SNI.
9. Fiber-PNC PN (MCC1 Media Gateway, SCC1 Media Gateway, or G650 Media Gateway stack [shown]) consisting of one or more media gateways or carriers. This PN has one TN2602AP circuit pack.
10. Control gateway or carrier in the A position in the stack. The control gateway or carrier contains:
    - A TN570Bv7/C/D EI circuit pack for bearer network connections to the SNI.

    NOTE: One TN2182 Tone Clock circuit pack must also be present per PN if the PN(s) consist of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312AP/BP IPSI circuit pack must be present per PN if the PN(s) consist of G650 Media Gateways.
11. TN 570Bv7/C/D to TN573 fiber connections between PNs and SNC
12. TN 573/570Bv7/C/D fiber connections between the SNCs and the A carrier (if the MCC1 is a PN)
13. Customer LAN
14. LAN connections of TN2302AP IP Media Interface, TN2602AP IP Media Resource 320, or TN799DP C-LAN for control of IP endpoints

    NOTE: The number of TN2302AP and TN799DP circuit packs varies, depending on the number of IP endpoints, port networks, and adjunct systems. These circuit packs may be inserted into a port carrier (shown in figure) or the PN control carrier.
15. LAN connections of media servers for remote administration
16. Duplicated server links, including the fiber link for translations transfer and the DAL2 link or software duplication link for control data sharing
17. IP-PNC PN (G650 Media Gateway or stack [shown in figure]). May also be an MCC1 from a DEFINITY Server migration or an SCC1.

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Figure notes: CSS-connected PNs (single control network) and IP-PNC PNs (duplicated control and duplicated bearer network) example (continued)

18. Control gateway or carrier, in the A position in the gateway stack, for PN 17. The control gateway contains:
    - A TN2312AP/BP IPSI circuit pack for IP connection to server.

      NOTE: For the G650 Media Gateway, the BP version of the TN2312 is required in order to provide environmental maintenance.
    - A TN2602AP IP Media Resource 320 for PN bearer connections over the LAN

      NOTE: The TN2602AP circuit pack may be placed in any gateway in the PN. However, the pair of TN2602AP circuit packs should be separated between two different gateways whenever possible.
19. Media gateway or carrier, in the B position in the gateway stack, , which contains:
    - A duplicated TN2312AP/BP IPSI circuit pack for duplicated control network to PN 17.
    - A duplicated TN2602AP IP Media Resource 320 for PN bearer connections over the LAN.

      NOTE: The TN2602AP circuit pack may be placed in any gateway in the PN. However, the pair of TN2602 circuit packs should be separated between two different gateways whenever possible.
20. LAN connections of duplicated TN2602AP IP Media Resource 320 circuit packs for IP-TDM voice processing. Connections to separate Ethernet switches are recommended, but not required.
21. Fiber-PNC PN (MCC1 Media Gateway, SCC1 Media Gateway, or G650 Media Gateway stack [shown]) consisting of one or more media gateways or carriers. This PN has no TN2602AP circuit packs and relies only on fiber connections for the bearer network.

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Example of combining IP- and ATM-connected PNs and different reliability levels

Figure 23 illustrates an S8710 Media Server configuration that combines ATM-connected PNs with standard duplex-server-only reliability and IP-PNC PNs with duplicated control network reliability.

Note:

In this example, the IP-PNC PN (item 22 in Figure 23) is connected to two Ethernet switches on the customer network since the ATM-connected PNs are connected to the server over a dedicated Ethernet switch.

Figure 23: Example of ATM-connect PNs with single control network and IP-PNC PNs with duplicated control network

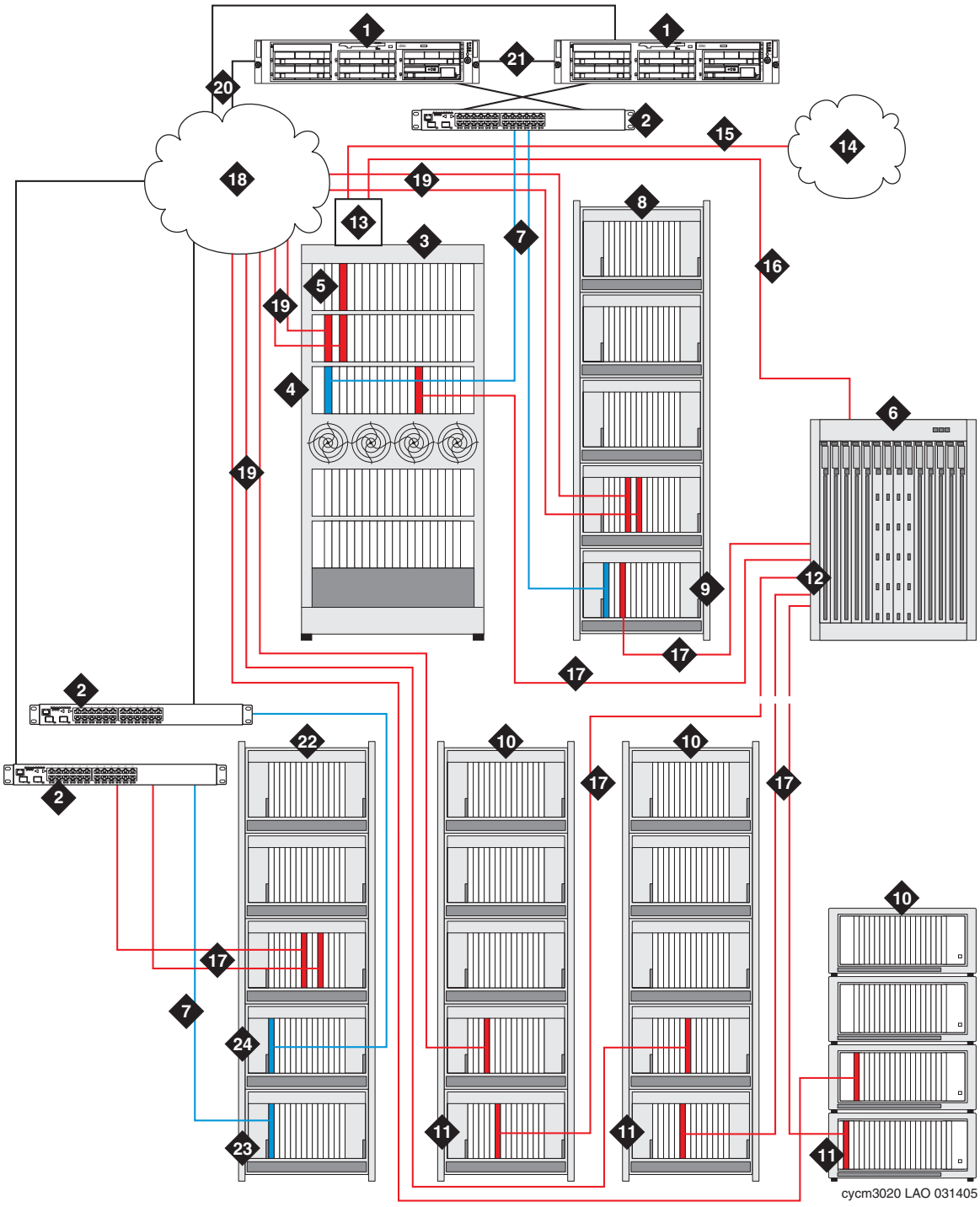


Figure notes: Example of ATM-connect PNs with single control network and IP-PNC PNs with duplicated control network

1. S8710/S8720 Media Server
2. Ethernet Switch
3. Fiber-PNC PN (MCC1 [shown], SCC1, or G650 Media Gateway)

   NOTE: A TN2302AP Media Interface or TN23602 Media Resource 320 for IP-TDM voice processing is required in at least one fiber-PNC PN for the combined PN connection methods to work.
4. Control carrier, in the A position in the MCC1, for PN 3. The control carrier contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.
   - A TN2305 or TN2306 circuit pack for bearer network connections to the ATM switch.
5. Carrier with TN464GP DS-1 circuit pack, for clock synchronization with a network resource
6. ATM switch.
7. IPSI-to-server control network connection via Ethernet switch
8. Fiber-PNC and server-connected PN (G650 Media Gateway stack [shown],MCC1 Media Gateway, or SCC1 Media Gateway stack, consisting of at least two media gateways or carriers).
9. Control gateway or carrier, in the A position in the stack, for PN 8. The control gateway contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.
   - A TN2305 or T2306 circuit pack for bearer network connections to the ATM switch.
10. Fiber-PNC PN (G650 Media Gateway stack (shown), MCC1 Media Gateway, SCC1 Media Gateway stack [shown]), consisting of at least two media gateways or carriers.
11. Control gateway or carrier, in the A position in the stack, for PN 10. The control gateway contains:
    - A TN2305 or TN2306 circuit pack for bearer network connections to the ATM switch.

    NOTE: One TN2182 Tone Clock circuit pack must also be present per PN if the PN(s) consist of SCC1 or MCC1 Media Gateways. One maintenance-only TN2312BP IPSI circuit pack must be present per PN if the PN(s) consist of G650 Media Gateways.
12. OC-3 connections to the ATM switch
13. Sync splitter
14. Public network (PSTN)
15. DS1 connection to sync splitter.
16. Timing signal to ATM switch from sync splitter.
17. Fiber connections from TN2305/TN2306 to ATM switch.
18. Customer LAN
19. LAN connections of TN2302AP Media Interface or TN2602AP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints
20. LAN connections of media servers for remote administration
21. Duplicated server links, including the fiber link for translations transfer and the link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.
22. IP-PNC PN (G650 Media Gateway or stack [shown in figure]). May also be an MCC1 from a DEFINITY Server migration or an SCC1.
23. Control gateway, in the A position in the gateway stack, for PN 22. The control gateway contains:
    - A TN2312AP/BP IPSI circuit pack for IP connection to server.
24. Media gateway, in the B position in the gateway stack, with duplicated TN2312AP/BP IPSI circuit pack for duplicated control network to server.

MCC1 Media Gateway with IP-PNC PNs or a combination of IP- and fiber-PNC PNs

An MCC1 Media Gateway may contain up to 5 PNs, with each carrier administered as a fiber-PNC PN. For migrations and conversions only to Communication Manager R3.0, an MCC1 can also support from 1 to 5 IP-PNC PNs, or both IP-PNC and fiber-PNC PNs. In this way, a combination of PN connection methods may exist on a single MCC1 Media Gateway.

An MCC1 may also contain up to two IP-PNC PNs with duplicated control networks. However, if a server-connected MCC1 uses duplicated bearer networks with CSS, such that switch node carriers must occupy the D and E positions on the MCC1, the MCC1 Media Gateway may house up to three PNs, but only one PN can have duplicated control.

The following tables identify the port network configuration options for IP-PNC and combined IP- and fiber-PNC PNs in an MCC1 Media Gateway.

Options for IP-PNC PNs in an MCC1 Media Gateway

The following diagrams indicate the PN options available using a single MCC1 Media Gateway with all-IP-PNC PNs. Each PN within the MCC1 Media Gateway is indicated by bold borders (**—**). Carriers within PNs are indicated by thin borders (—).

	MCC1 with 1 PN with single control	**MCC1 with 1 PN with duplicated control**	**MCC1 with 2 PNs with single control**	**MCC1 with 2 PNs one with duplicated control**	**MCC1 with 2 PNs both with duplicated control**
C Carrier					
B Carrier		IPSI (secondary)			IPSI (secondary)
A Carrier	IPSI	IPSI (primary)	IPSI	IPSI	IPSI (primary)
D Carrier				IPSI (secondary)	IPSI (secondary)
E Carrier			IPSI	IPSI (primary)	IPSI (primary)

	MCC1 with 3 PNs with single control	MCC1 with 3 PNs one with duplicated control	MCC1 with 4 PNs with single control	MCC1 with 4 PNs one with duplicated control	MCC1 with 5 PNs with single control
C Carrier			IPSI	IPSI	IPSI
B Carrier	IPSI	IPSI	IPSI	IPSI	IPSI
A Carrier	IPSI	IPSI	IPSI	IPSI	IPSI
D Carrier	IPSI	IPSI (secondary)	IPSI	IPSI (secondary)	IPSI
E Carrier		IPSI (primary)		IPSI (primary)	IPSI

Options for combined IP- and fiber-PNC PNs in an MCC1 Media Gateway (single control network)

The following diagrams indicate the PN options available using a single MCC1 Media Gateway with IP-PNC PNs, fiber-PNC (direct, CSS, or ATM-connected) PNs, and single control networks. Where "fiber-PNC" is indicated, the PN may contain an IPSI for a connection to the server or may only contain expansion interface circuit packs for fiber connections to other PNs. Each PN within the MCC1 Media Gateway is indicated by bold borders (**—**). Carriers within PNs are indicated by thin borders (—).

	MCC1 with 2 PNs	MCC1 with 3 PNs with single control	MCC1 with 4 PNs with single control	MCC1 with 5 PNs with single control
C Carrier			IP-PNC or fiber-PNC	IP-PNC or fiber-PNC
B Carrier		IP-PNC or fiber-PNC	IP-PNC or fiber-PNC	IP-PNC or fiber-PNC
A Carrier	IP-PNC or fiber-connected	IP-PNC or fiber-PNC	IP-PNC or fiber-PNC	IP-PNC or fiber-PNC
D Carrier		IP-PNC or fiber-PNC	IP-PNC or fiber-PNC	IP-PNC or fiber-PNC
E Carrier	IP-PNC or fiber-PNC			IP-PNC or fiber-PNC

Options for combined IP- and fiber-PNC PNs in an MCC1 Media Gateway (duplicated control networks)

The following diagram indicates the PN options available using a single MCC1 Media Gateway with IP-PNC PNs, fiber-PNC (direct, CSS, or ATM-connected) PNs, and duplicated control networks. Each PN within the MCC1 Media Gateway is indicated by bold borders (**—**). Carriers within PNs are indicated by thin borders (—).

Note:

The configurations in the following diagram assume the bearer network for the fiber-PNC PNs is *not* duplicated. For configurations with duplicated bearer networks, see Options for combined IP- and fiber-PNC PNs in an MCC1 Media Gateway (duplicated control and bearer networks) on page 97.

	MCC1 with 2 PNs one with duplicated control	**MCC1 with 2 PNs one with duplicated control[1]**	**MCC1 with 2 PNs both with duplicated control**	**MCC1 with 3 PNs one with duplicated control**	**MCC1 with 4 PNs one with duplicated control**
C Carrier					IPSI for IP-PNC or fiber-PNC with no IPSI
B Carrier	IPSI for IP-PNC or IPSI for fiber-PNC (secondary)		IPSI for IP-PNC (secondary)	IPSI for IP-PNC or fiber-PNC with no IPSI[1]	PSI for IP-PNC or fiber-PNC with no IPSI
A Carrier	IPSI for IP-PNC or IPSI for fiber-PNC (primary)	IPSI for IP-PNC, IPSI for fiber-PNC, or fiber-PNC with no IPSI	IPSI for IP-PNC (primary)	IPSI for IP-PNC or fiber-PNC with no IPSI[1]	IPSI for IP-PNC or fiber-PNC with no IPSI
D Carrier		IPSI for IP-PNC or IPSI for fiber-PNC (secondary)	IPSI for IP-PNC (secondary)	IPSI for IP-PNC (secondary)	IPSI for IP-PNC (secondary)
E Carrier	IPSI for IP-PNC, IPSI for fiber-PNC, or fiber-PNC with no IPSI	IPSI for IP-PNC or IPSI for fiber-PNC (primary)	IPSI for IP-PNC (primary)	IPSI for IP-PNC (primary)	IPSI for IP-PNC (primary)

1. If the PN using Carriers D and E is fiber-PNC with duplicated IPSIs, Carriers A, B, and C may not contain IP-PNC PNs.

Options for combined IP- and fiber-PNC PNs in an MCC1 Media Gateway (duplicated control and bearer networks)

The following diagram indicates the PN options available using a single MCC1 Media Gateway with an IP-PNC PN with a duplicated control network and a fiber-PNC (direct, CSS, or ATM-connected) PN with duplicated control and bearer networks. Each PN within the MCC1 Media Gateway is indicated by bold borders (**—**). Carriers within PNs are indicated by thin borders (—).

Note:

In the following illustration, the ISPIs that enable a duplicated control network for the fiber-PNC PN reside in another fiber-PNC PN.

	MCC1 with 2 PNs one with duplicated control and bearer network[1]	**MCC1 with 2 PNs one with duplicated control and bearer network**[1]
C Carrier		
B Carrier	IPSI for IP-PNC (secondary)	Fiber-PNC with no IPSI (secondary)
A Carrier	IPSI for IP-PNC (primary)	Fiber-PNC with no IPSI (primary)
D Carrier	Fiber-PNC with no IPSI (secondary)	IPSI for IP-PNC (secondary)
E Carrier	Fiber-PNC with no IPSI (primary)	IPSI for IP-PNC (primary)

1. Duplicated bearer only available with PN that is fiber-PNC. Duplicated control exists in a different IPSI-connected PN.

Example of MCC1 IP-PNC

Figure 24 illustrates an S8700-series Media Server configuration that uses the carriers in an MCC1 Media Gateway as IP-PNC PNs. This configuration is available with a migration from a DEFINITY Server SI or R or a conversion from fiber-PNC to IP-PNC only. The example shows one PN with duplicated IPSIs (item 5) and two PNs (items 4 and 6) with single IPSIs, one consisting of a single carrier and the other with two carriers.

Figure 24: MCC1 Media Gateway with carriers as IP-PNC PNs (duplicated control network) example

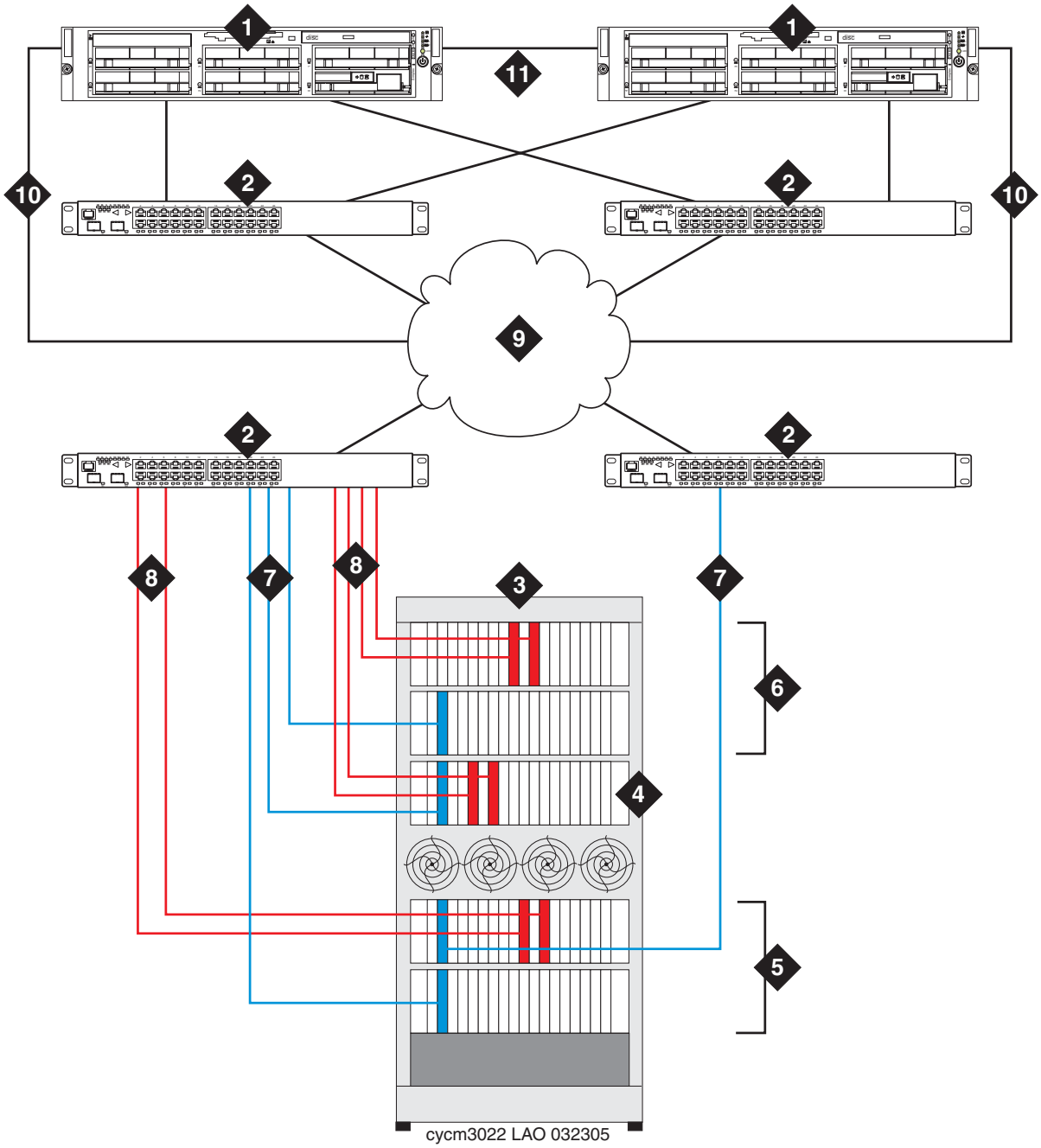


Figure notes: MCC1 Media Gateway with carriers as IP-PNC PNs (duplicated control network) example

1. S8700-series Media Server
2. Ethernet Switch
3. MCC1 Media Gateway
4. IP-PNC PN, with one expansion port carrier in the A position, which contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.
   - A TN2302AP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing
   - An optional TN799DP C-LAN for control of IP endpoints
5. IP-PNC PN, with two carriers, which contains:
   - E-position port carrier
     - A TN2312AP/BP IPSI circuit pack for IP connection to server.
   - D-position port carrier
     - A TN2312AP/BP IPSI circuit pack for IP connection to server.
     - A TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing
     - An optional TN799DP C-LAN for control of IP endpoints
6. IP-PNC PN, with two carriers, which contains:
   - C-position port carrier
     - A TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing
     - An optional TN799DP C-LAN for control of IP endpoints
   - B-position port carrier
     - A TN2312AP/BP IPSI circuit pack for IP connection to server.
7. IPSI-to-server control network connection via Ethernet switch
8. LAN connections of TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints

   NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, port networks, and adjunct systems. These circuit packs may be inserted into a port carrier (shown in figure) or the PN control carrier.
9. Customer LAN
10. LAN connections of media servers for remote administration
11. Duplicated server links, including the fiber link for translations transfer and the DAL2 link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.

Example of MCC1 with IP- and fiber-PNC PNs

Figure 25 illustrates an S8700-series Media Server configuration that uses the carriers in an MCC1 Media Gateway as both fiber-PNC and IP-PNC PNs. The MCC1 Media Gateway (item 3) contains two IP-PNC PNs (items 5 and 6) with a third CSS-connected PN consisting of a single carrier (item 4). With a TN2602AP IP Media Resource 320 or TN2302AP IP Media Processor, the CSS-connected PN serves as a gateway between the IP-PNC PNs and the fiber-PNC PNs. Note that the MCC1 Media Gateway also contains a CSS or Switch Node Carrier (SNC).

Figure 25: MCC1 Media Gateway with IP- and fiber-PNC PNs example

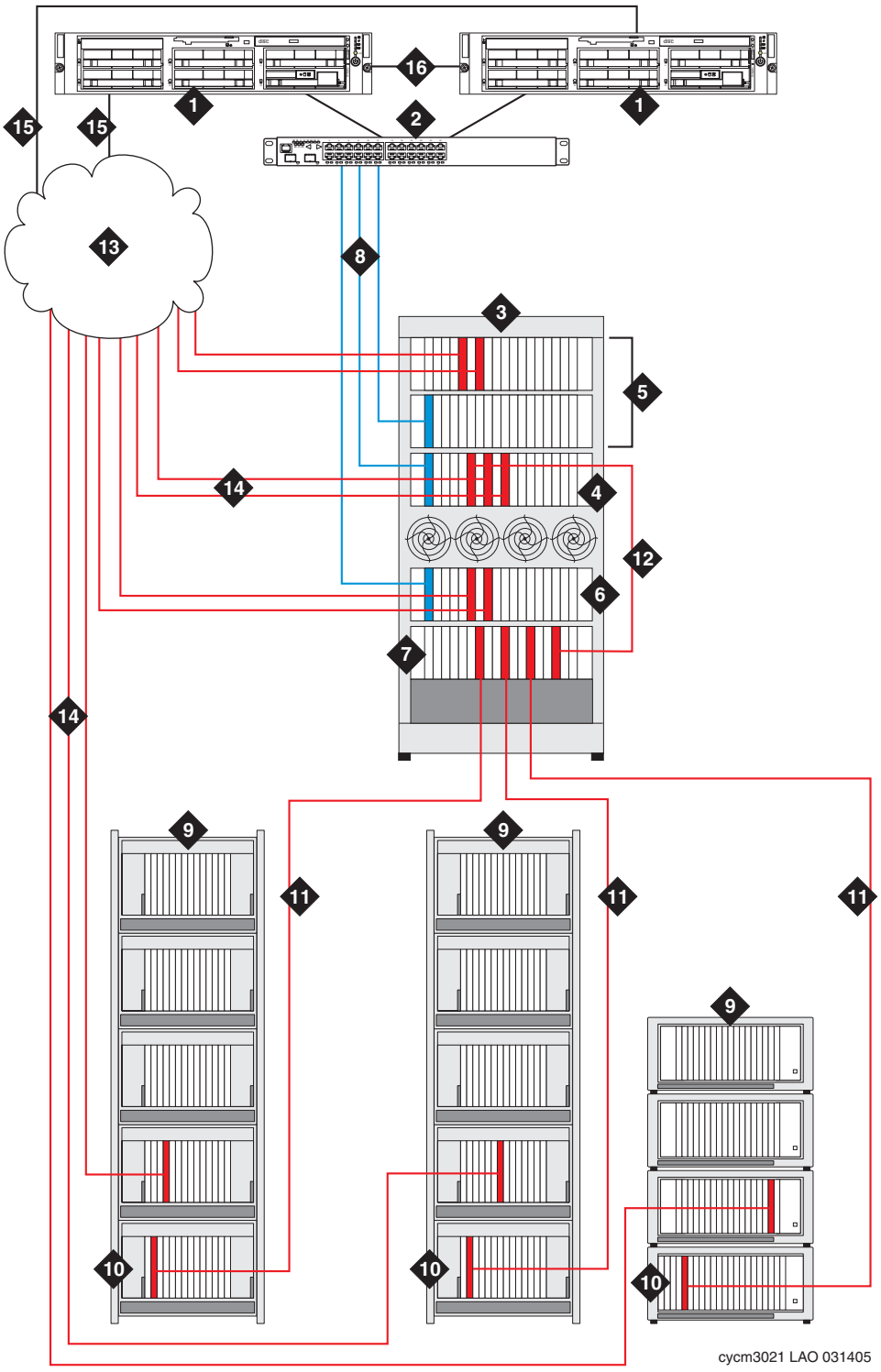


Figure notes: MCC1 Media Gateway with IP- and fiber-PNC PNs

1. S8700-series Media Server
2. Ethernet Switch
3. MCC1 Media Gateway (CSS and PN)
4. CSS-connected PN carrier, in the A position, which serves as a gateway to IP-PNC PNs. The PN contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.
   - A TN570Bv7/C/D EI circuit pack for bearer network connections to the Switch Node Carrier (SNC).
   - A TN2302AP IP Media Processor or TN2602AP IP Media Resource 320 circuit pack. These circuit packs enable the PN to be a gateway between the fiber-PNC and IP-PNC PNs.
   - An optional TN799DP C-LAN circuit pack for control of IP endpoints
5. IP-PNC PN, consisting of carriers in the B and C positions. Carrier B contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server. The bottom carrier in the PN must contain the primary IPSI circuit pack.

   Carrier C contains:
   - A TN2302AP IP Media Processor or TN2602AP IP Media Resource 320 circuit pack.
   - An optional TN799DP C-LAN circuit pack for control of IP endpoints

     These circuit packs can actually be inserted in any carrier within the PN.
6. IP-PNC PN, consisting of one carrier in the D position. Carrier D contains:
   - A TN2312AP/BP IPSI circuit pack for IP connection to server.
   - A TN2302AP IP Media Processor or TN2602AP IP Media Resource 320 circuit pack.
   - An optional TN799DP C-LAN circuit pack for control of IP endpoints
7. Switch node carrier (SNC) or CSS, which contains:
   - Multiple TN573 SNI circuit packs for EI connections to PNs
8. IPSI-to-server control network connection via Ethernet switch
9. CSS-connected PN (G650 Media Gateway or stack [shown in figure], MCC1 Media Gateway, or SCC1 Media Gateway stack [shown in figure]).
10. Control gateway or carrier, in the A position in the stack, for PNs labeled 9. The control gateway contains:
    - A TN2312AP/BP IPSI circuit pack for IP connection to server.
    - A TN570Bv7/C/D EI circuit pack for bearer network connections to the SNI.
11. TN 570Bv7/C/D to TN573 fiber connections between PNs and SNC
12. TN 573/570Bv7/C/D fiber connections between the SNCs and the B carriers (if the MCC1 is a PN)
13. Customer LAN
14. LAN connections of TN2302AP IP Media Interface or TN2602AP IP Media Resource 320 for IP-TDM voice processing and optional TN799DP C-LAN for control of IP endpoints

    NOTE: The number of TN2302AP, TN2602AP, and TN799DP circuit packs varies, depending on the number of IP endpoints, port networks, and adjunct systems. These circuit packs may be inserted into a port carrier or the PN control carrier.
15. LAN connections of media servers for remote administration
16. Duplicated server links, including the fiber link for translations transfer and link for control data sharing. The link for control data sharing is implemented through the DAL2 board or (for the S8720 media server) through software duplication.

ESS support for combined IP- and fiber-PNC PNs

Any Enterprise Survivable Server (ESS) can also support a combined IP- and fiber-PNC configuration in the event of failover to the ESS. Both an S8500/S8500B/S8500C and an S8700-series ESS can support single control and duplicated control networks for both the IP-PNC and fiber-PNC portions of the configuration, However, the ESSs can support only those CSS- or ATM-connected PNs that individually have a TN2312AP/BP IPSI circuit pack and either a TN2302AP IP Media Processor or TN2602AP IP Media Resource 320 circuit pack. This limitation exists because the ESS provides only IP-PNC control and bearer service to PNs.

For more information on ESS, see the *Using the Avaya Enterprise Survivable Servers (ESS)*, 03-300428.

Chapter 2: Control Networks for S8700-Series and S8500 Media Servers

Control networks are the networks over which media servers, such as the S8700-series or S8500 Media Servers, exchange signaling data with the port networks through the IPSI circuit packs.

This chapter provides information on how to set up control networks. Topics covered include:

Control network C

Control network C (CNC) was introduced in Avaya Communication Manager 3.0. It allows control connectivity to be passed through the customer network interface. This functionality is introduced to simplify the network design for enterprises with local private control networks (control network A and control network B) who wish to use their corporate network to support remote IPSI-controlled port networks.

Control network C functionality is useful in situations where an enterprise is adding distributed port networks at remote sites connected to a centralized S8700-series or S8500 server. Using control network C allows the enterprise to keep CNA and CNB on a private network while other port networks communicate remotely. This can help maintain the security and reliability of the existing port networks connected to control networks A and B. New port networks can still connect to the media server without extending control networks A and B to remote sites nor requiring the use of static routes on the S8700-series or S8500 media servers.

Important:

Control network C is a *server* enhancement. control network C could be used in an all IP-connected scenario, as well as a scenario in which fiber-connected and IP-connected port networks are connected in a single configuration.

To enable, disable or report the current status for control network C on an Avaya S8700-series or S8500 Media Server, use the graphical maintenance web interface.

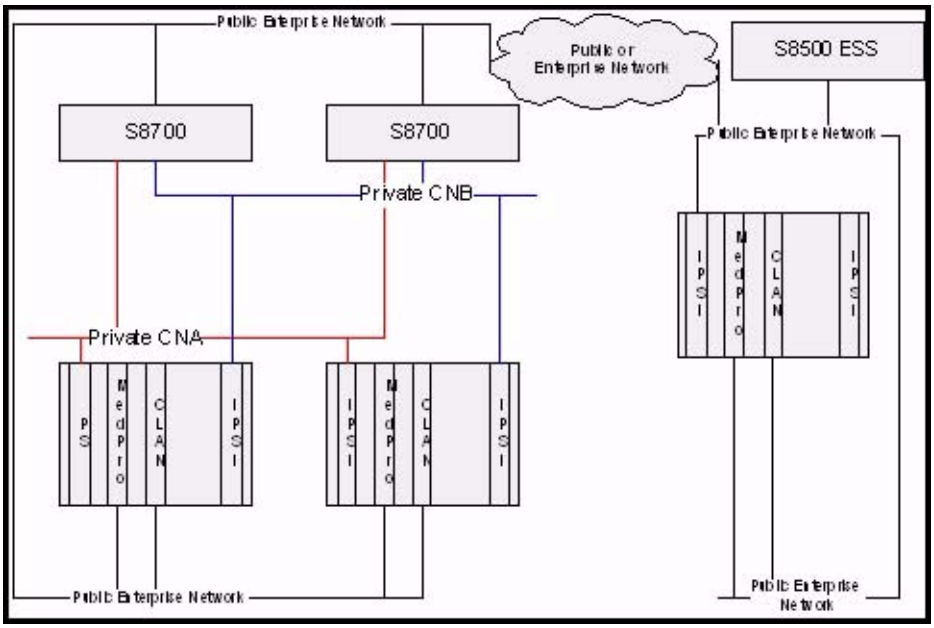
Avaya recommends that port networks be attached to private control networks A and B within a building, but that remote port networks connect to the media servers through control network C. This offers protection against network disruptions and Denial of Service (DoS) attacks to Port

Networks in the central site, while offering flexibility and reducing costs when attaching port networks at remote sites.

CNC configuration: Multi-site private CNA, CNB, with remote PNs on public LAN

This example shows the connection of local private control networks using the existing public enterprise network to provide connectivity to a remote site with an IPSI-controlled port network and an S8500 ESS server. The local control networks are designated as private in this case because the IP addressing of these control networks will not be routable through the enterprise network. The control network at the remote site is designated as public because it is fully routable throughout the enterprise network. The Control connection from the S8700 to the remote IPSI is established through the “Customer LAN”, or the third interface connected to the enterprise network. This configuration is particularly appropriate for large main sites, which require a fully redundant architecture, with smaller remote sites that do not require the same level of redundancy.

This design provides for total protection of the local control networks from any enterprise network failures; however, the remote site may be affected by enterprise network issues. Configuration is simplified because the default route of the CNC interface allows the CNC interface to communicate across the enterprise routed network infrastructure without requiring static routes.



Advantages: - The dedicated control network provides total isolation from outages in the enterprise network, so all local TDM communication at the main site can remain active during total enterprise network failure. There are no static routes to maintain.

Disadvantages: - The remote site can be affected by public enterprise network issues. The remote ESS server cannot control the port networks at the main site.

Combining fiber-connected and IP-connected port networks in a single configuration

With Communication Manager 3.0 and later, Avaya extends “Control Network on Customer LAN” functionality to simplify network configuration by allowing both fiber-connected and IP-connected port networks in a single configuration. With combined port network functionality, enterprises can attach IP-connected, ATM-connected, or center-stage-connected port networks to their S8700-series media server. Likewise, they can attach IP-connected or fiber-connected PNs to their S8500 media server(s).

To support combined port networks, Avaya has enhanced the flexibility of control networks for port network attachment. In addition to private control networks A and B, Avaya allows the “Customer LAN” Ethernet interface to be used as a third, public control network, Control network C.

Sample configurations

Network connectivity between S8700-series servers and port networks

The Avaya S8700 solution requires IP connectivity between S8700-series interfaces and Avaya media gateways. IP-connected port networks use IPSI cards in the port networks to communicate with the Media Server. This connection will be referred to as the “Control Connection”. There are many network options to provide this connectivity, and it is at the enterprise’s discretion how this is best implemented in its environment.

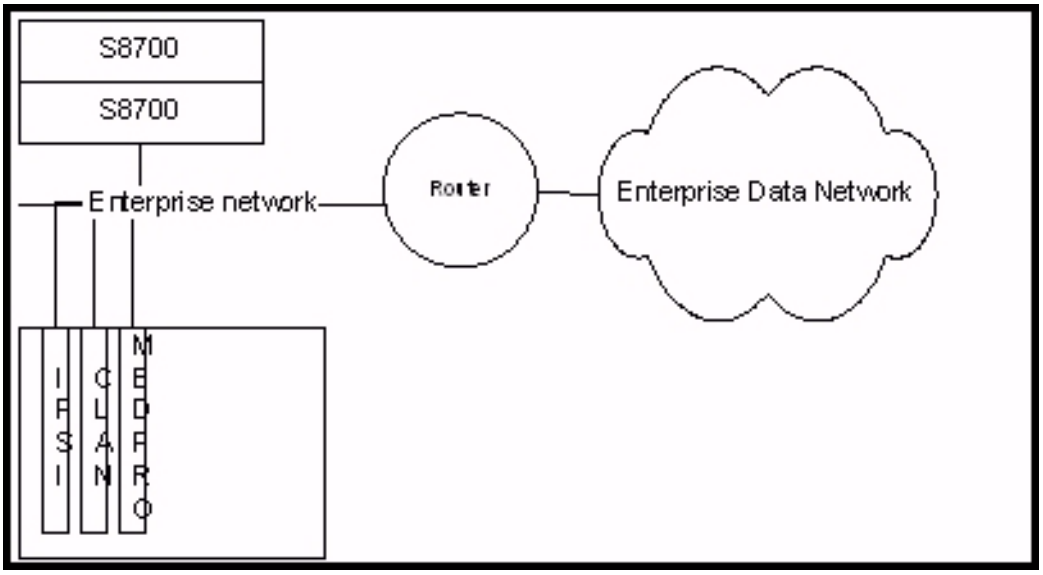
If IP connectivity, including the control connection, between the server and port network is lost, the server will be unable to provide call control, resulting in an unstable system. Although the Avaya S8700-series media server interfaces provide for Denial of Service protection, they cannot affect the ability of the network to successfully forward packets during a virus or worm attack, or when the network becomes unstable due to network outages or administrative errors.

In hybrid environments (such as, IP and TDM endpoints and trunks), the incentive to minimize disruption of the IP control connection is increased. By maintaining the control connection when other network components have failed, TDM-connected endpoints will continue to function.

The following examples illustrate common methods for designing the control connection between S8700-series servers and IP-connected port networks. They identify advantages and disadvantages of each, so enterprises can select the appropriate solution for their environment.

Example 1: IP-connected, single-site, single subnet

This design connects all Avaya server and gateway interfaces to a single VLAN. This solution is used primarily in small sites of less than 500 users.

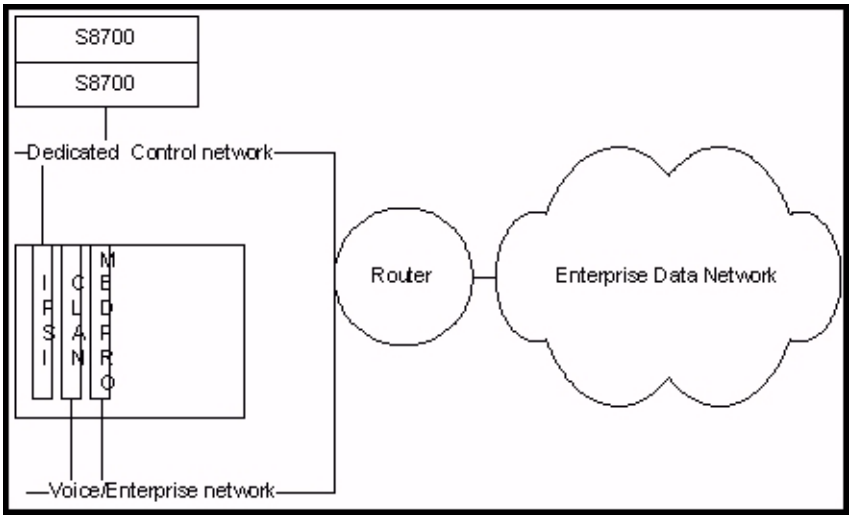


Advantages: - Simple; no host-based static routing required.

Disadvantages: - Provides no control point to protect the “control connection” from network conditions that would not allow IP packets to reach their destinations. Because endpoints on the enterprise data Network (even if given a separate “Voice VLAN”) must access C-LANs and Media Processors using a large variety of ports, the control connection can be negatively affected by DoS attacks, viruses, network convergence events, and so on.

Example 2: IP-connected, single-site, with a dedicated "control" network

This design connects all Avaya servers and IPSIs to a dedicated control network. C-LANS and Media Processors are connected to a separate voice VLAN. Additional separation from the infrastructure can be achieved by using a separate isolated switch for the dedicated control network, providing resiliency from spanning tree calculations and DoS attacks that could potentially disrupt a switch connected to the enterprise infrastructure. This design is typical in large single site deployments. Firewalls are often used to provide additional security.

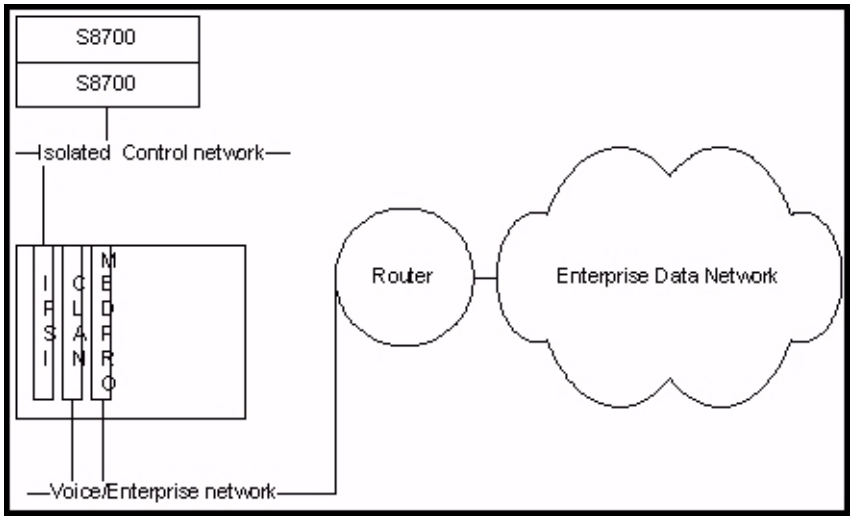


Advantages: - Provides a control point to limit traffic allowed on the control network. An additional switch can provide protection against enterprise network failures. No host-based static routing required.

Disadvantages: - Requires an additional VLAN or dedicated switch/router interface.

Example 3: IP-connected, single-site, with an isolated "control" network

An isolated control network provides little value if the isolation is through the use of VLANs only. A switch not connected to any network infrastructure will provide full protection form external attack. It is still possible to administer the Avaya Communication Manager server through a properly configured C-LAN connected to the enterprise network. This design is not common, but is used by some enterprises to provide total isolation of the control network.

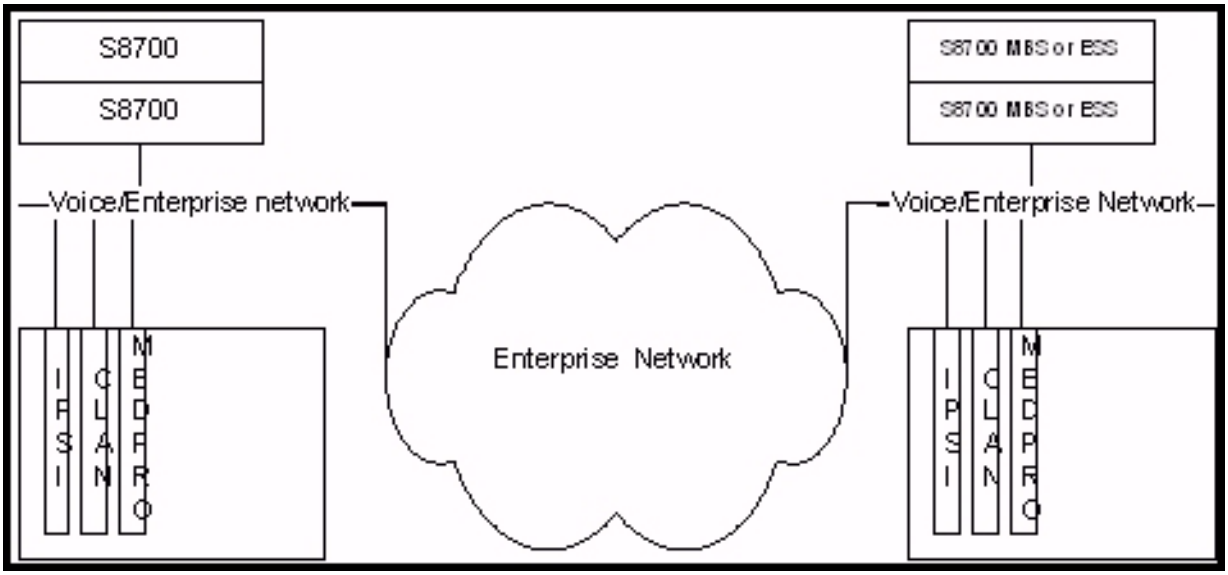


Advantages: - Provides total isolation of the control network. No host-based static routing required.

Disadvantages: - Requires an additional switch. The user cannot access the Web interface from the enterprise network. The user must configure a C-LAN card to accept administration connections.

Example 4: IP-connected, multi-site, single subnet, with a backup cluster/ESS

This design connects all Avaya server and gateway interfaces to a single VLAN per location. It is important to note that for the primary cluster to control the port networks at the remote site, the primary servers must have IP connectivity to the remote IPSIs. Also, for the backup cluster to take control of the primary sites port networks, it must have IP connectivity to the primary site IPSIs across the network. This design is not often used. Most large sites have chosen to separate the control network for increased reliability.



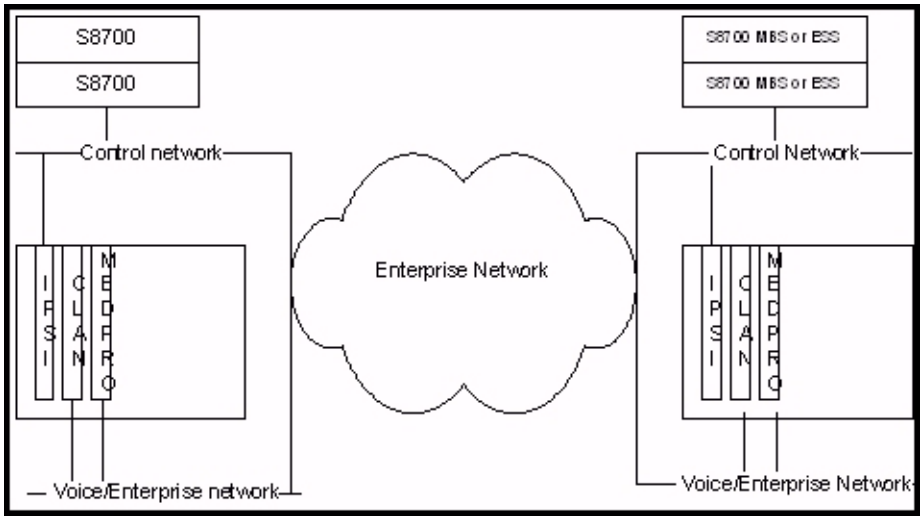
Advantages: - Simple; no host-based static routing required.

Disadvantages: - Provides no control point to protect the “control connection” from network conditions that would not allow IP packets to reach their destinations. Using this design, the control connection can be negatively affected by DoS attacks, viruses, spanning tree calculations, and so on. Any disruption in IP connectivity will also disrupt the TDM connections.

Example 5: IP-connected, multi-site, with a dedicated routed "control" network

The above example shows two sites: the main site with the primary server cluster, and a remote site with a backup cluster. To provide protection of the Server-to-IPSI link, Avaya recommends the use of a dedicated control network. For backup cluster redundancy, it is a requirement that each server pair be able to communicate across the enterprise network to control remote port networks.

It is not a requirement, nor is it recommended that the voice (or data) networks be able to communicate using the control networks. It is recommended that strong access lists or a firewall separate the voice and data networks from the control network to limit traffic allowed from the outside networks. Tight control of the rule set can then allow for specific stations to access the web interface of the S8700 Servers. Once again, the IP control connection must be permitted through any access lists or firewalls. Control connectivity is required between each server cluster and the IPSIs of any port network they wish to control. This is the most prevalent design in large corporate infrastructures supporting the Avaya S8700-series IP Telephony Solutions.



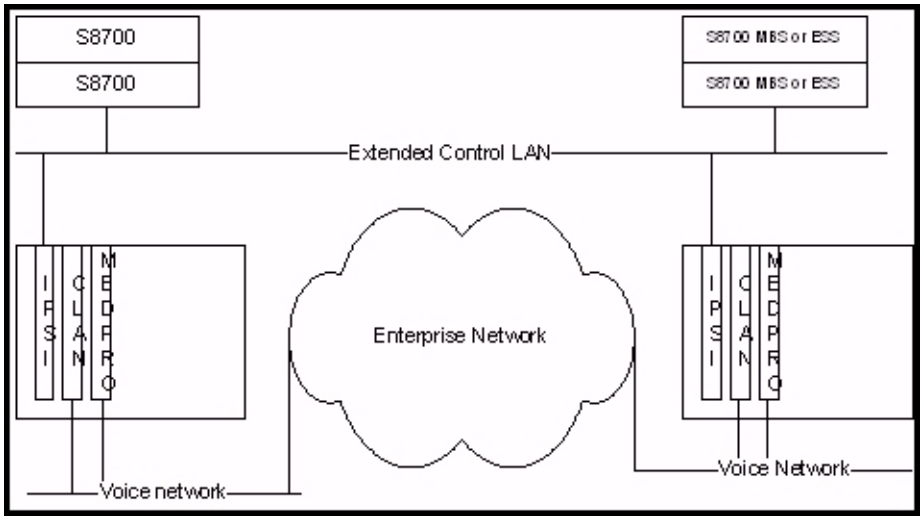
Advantages: - Provides a control point to limit traffic allowed on the control network. With additional Ethernet switches, it can provide protection against utilization failures and spanning tree recalculations. This design can allow TDM connections to continue during specific network failures. No host-based static routing required.

Disadvantages: - Requires additional VLANs and/or dedicated switches and router interfaces.

Example 6: Multi-site with a dedicated extended Layer 2 "control" network

This example shows the use of a single extended VLAN providing Layer 2 connectivity between sites. This design provides all the benefits of design #5 and also addresses resiliency of the enterprise network failing at Layer 3. It is at the enterprise's discretion to route the traffic on the extended control LAN to the enterprise network to provide access for administrative functions.

This design has been used successfully in several large-scale, Avaya IP-connected deployments. It provides excellent reliability, especially when used with redundant network equipment, but is expensive and some times impossible due to fiber-optic cable availability and other network design consideration between the sites.



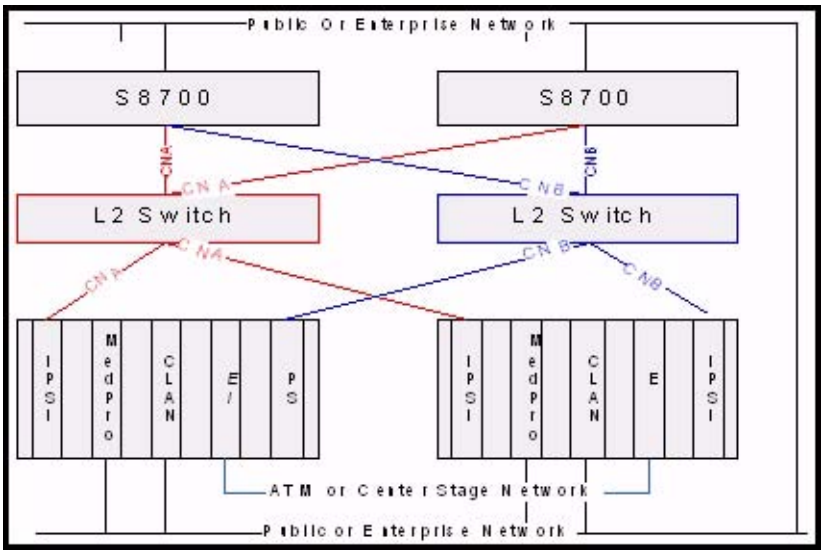
Advantages: - Provides a control point to limit traffic allowed on the control network. With additional switches, it can provide protection against switch failures and spanning tree recalculations. This will allow TDM connections to continue during most network failures. No host-based static routing required.

Disadvantages: - Requires additional dedicated switches, and a dedicated physical connection infrastructure.

Example 7: Single-site, fiber-connected or IP-connected, with redundant control interfaces

The fiber-connected (formally Multi-Connect offer) configuration had several choices for reliability. Two offers provided redundant servers and interfaces on two private control networks. Administrative control is provided by an interface directly on the enterprise (“public”) network, or through properly administered C-LANs. For the purposes of this document, public network refers to the routed enterprise network, and not necessarily networks capable of being routed on the Internet.

The fiber-connected (formerly Multi-Connect) configurations are distinguished by the existence of a non-IP bearer path between port networks as shown in the figure.



Advantages: - Provides total isolation of the private control networks. This design allows TDM connections to continue during any single control network component failure. No host-based static routing is required.

Disadvantages: - Requires additional switches. It cannot extend across a routed infrastructure.

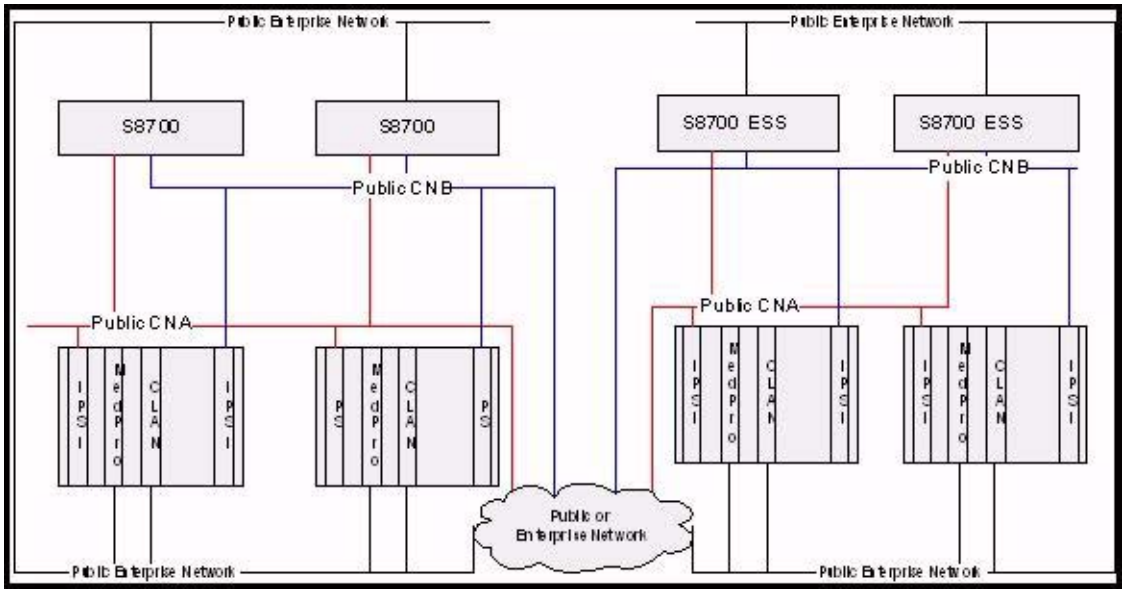
Control network on customer LAN (CNOCL)

Avaya Communication Manager 2.0 introduced the control network on Customer LAN option, which allows the use of routed control networks. CNOCL removed many of the IP connectivity differences between an IP-connected and fiber-connected (formally Multi-Connect), and leaves the only true difference being the existence of inter-port network bearer paths. CNOCL provides enterprises with several options to create and extend control networks

Example 8: Multi-site CNOCL using merged enterprise and control network

This example shows the connection of the two private control networks to the customers enterprise network, making them public. They are designated public in this case because the IP addressing of these control networks must be routable through the enterprise network.

This design has been used successfully in several Avaya deployments, but opens the control networks to all network issues experienced in the enterprise. Firewalls or strong access lists should be used to protect each site's control network, but inter-site connectivity cannot truly be protected. The use of the third interface connecting to the enterprise infrastructure for management is no longer necessary, and can be collapsed on the one of the other two networks.



Advantages: - Provides a control point to limit traffic allowed on the control network. Uses the enterprise's existing network infrastructure.

Disadvantages: - This will not allow TDM connections to continue during most network failures. Static routing is required on both Main and MBS/ESS servers, and may become complex, depending on the network architecture. Changes in network architecture will have to be synchronized with changes in the static route table, and will be service-affecting.

Example 9: Multi-site CNOCL using extended private networks

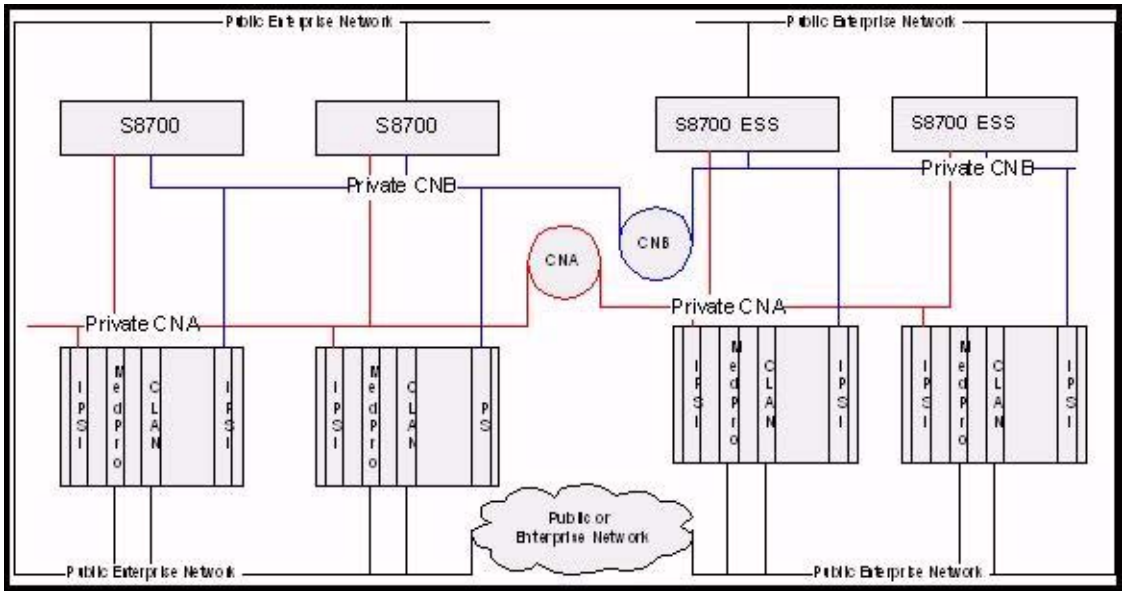
This example shows the connection of the two private control networks using a dedicated routed infrastructure. They are designated private in this case because the IP addressing of these control networks is not routable through the enterprise network.

This design provides for total protection of the control networks from any enterprise network failures. With proper architecture, the static routing for CNA and CNB can be reduced to single summary routes, rather than static routes per IPSI.

Example:

route 192.168.0.0 255.255.128.0 CNA

route 192.168.128.0 255.255.128.0 CNB



Advantages: - The dedicated Control network provides total isolation from outages in the enterprise network, so all TDM communication can remain active during total enterprise network failure. The use of simple summary routes instead of possibly complex static routing provides for a more reliable system. The synchronization of network changes with Communication Manager can be logistically difficult.

Disadvantages: - Requires a dedicated infrastructure.

Chapter 3: Administering converged networks

This section provides information for administering converged network components.

About Voice over IP converged networks

Until recently, voice, video, and data were delivered over separate, single-purpose networks. A converged network brings voice, data, and video traffic together on a single IP network. Avaya's VoIP technology provides a cost-effective and flexible way of building enterprise communications systems through a converged network.

Some of the flexible elements of a converged network include:

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

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

Providing a network assessment

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

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

- Basic Network Assessment — is a high-level LAN/WAN infrastructure evaluation that determines the suitability of an existing network for VoIP.
- Detailed Network Assessment — is typically the second phase in the Network Assessment for IP Telephony solutions.

  The detailed network assessment takes information gathered in the basic network assessment, performs problem diagnosis, and provides functional requirements for the network to implement Avaya VoIP.

For more information, see

- "Network assessment offer" in *Avaya Application Solutions: IP Telephony Deployment Guide*, 555-245-600.
- Avaya Communication Solutions and Integration (CSI)

  Avaya Communication Solutions and Integration (CSI) supports a portfolio of consulting and engineering offers to help plan and design voice and data networks, including:

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

  You can contact Avaya CSI:

  - On the Web -- http://csi.avaya.com.
  - by E-Mail: bcsius@avaya.com
  - by phone: +1 866 282 9266
- http://netassess.avaya.com for a description of the Avaya network assessment policy. Note: this link is available only from within the Avaya corporate network.

Setting up VoIP hardware

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

- TN464HP/TN2464CP Universal DS1 circuit packs and MM710 T1/E1Media Module
- TN799DP Control LAN
- TN2302AP IP Media Processor
- TN2302AP IP Media Processor
- TN2602AP IP Media Resource 320
- TN2312BP IP Server Interface (IPSI)
- MM760 VoIP Media Module
- TN8400AP Media Server circuit pack
- TN8412AP S8400 server IP Interface

TN464HP/TN2464CP Universal DS1 circuit packs and MM710 T1/E1Media Module

The TN464HP/TN2464CP circuit packs and the MM710 Media Module (version 3 and later) have the same functionality as other DS1 circuit packs with the addition of echo cancellation circuitry, which offers echo cancellation tail lengths of up to 96 milliseconds (ms). The TN574, TN2313, and TN2464 DS1 circuit packs do not support echo cancellation.

The TN464HP/TN2464CP and MM710 are intended for users who encounter echo over circuits connected to the Direct Distance Dialing (DDD) network. Echo is most likely to occur when Avaya Communication Manager is configured for ATM, IP, and wideband. In addition, echo can occur on system interfaces to local service providers that do not routinely install echo cancellation equipment in all their circuits.

Echo cancellation is a software right-to-use feature that supports voice channels, and is not intended for data. When a data call is received, these circuit packs detect a modem tone and turn off echo cancellation for the duration of the data call.

Working with echo cancellation

You can determine whether echo cancellation is enabled for TN464HP/TN2464CP circuit packs and MM710 T1/E1 Media Modules by displaying the **system-parameters customer-options** screen.

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.

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

Note:

Any changes made to the echo cancellation settings on the DS1 Circuit Pack screen take effect immediately.

The **DS1 Circuit Pack** screen for the TN464HP/TN2464CP circuit packs and MM710 media module has fields to support echo cancellation: **Echo Cancellation**, **EC Direction**, and **EC Configuration**. The **Echo Cancellation** field appears when the Echo Cancellation feature is activated on the **System-Parameters Customer Options** screen. The **EC Direction** and **EC Configuration** fields appear when the **DS1 Echo Cancellation** field is enabled.

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

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

To administer the DS1 circuit pack and MM710 media module

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

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

DS1 Circuit Pack screen

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

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

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

                                      T303 Timer (sec): ___

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

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

Field	Conditions/Comments
Echo Cancellation	Enter **y** to enable echo cancellation on the Universal DS-1 circuit pack.
EC Direction	Indicates the direction of the echo that is being cancelled. Enter **inward** or **outward**. • The **inward** setting cancels echo energy coming back into the switch — energy from an outgoing call is reflected from an external reflection point (party "inside" the switch hears the echo). • The **outward** setting cancels echo energy going outside the switch — energy from an incoming call is reflected from an internal reflection point (party "outside" the switch hears the echo).

Field	Conditions/Comments
EC Configuration	Indicates the set of echo cancellation defaults to administer. Appears when the Echo Cancellation field is set to **y**. Enter digits between **1-15**. • Enter **1** or **5-15** to provide most rapid adaptation in detecting and correcting echo at the beginning of a call, regardless of the loudness of the talker's voice. For very loud talkers and severe echo, the far-end talker's speech is heard as clipped when both parties talk at the same time. • Enter **2** for slightly slower adaptation to echo, use if speech is often clipped when both parties talk at the same time. • Enter **3** for slightly slower adaptation to echo, may result in a 2 or 3 second fade on strong echo for quiet talkers. Completely removes speech clipping. • Enter **4** in cases of extreme echo, excessive clipping or breakup of speech. May result in slight echo or background noise. **Note:** For the MM710, the values **1** and **4** are reversed. That is, **1** for the MM710 is the same as **4** for the TN464HP/TN2464CP, and **4** for the MM710 is the same as **1** for the TN464HP/TN2464CP

Administering echo cancellation on trunks

Note:

Changes to echo cancellation settings on the Trunk Features screen do not take effect until after a port or trunk group is busied-out/released, or the SAT command **`test trunk group`** is performed, or periodic maintenance is run.

Echo cancellation is turned on or off on a per trunk-group basis using the **`change trunk-group`** command. If the trunk group field, **DS1 Echo Cancellation** is `y`, echo cancellation is applied to every TN464HP/TN2464CP trunk member in that trunk group. The echo cancellation parameters used for a given trunk member are determined by the **EC Configuration** number administered on the **DS1 Circuit Pack** screen for that specific trunk's board.

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

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

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

To administer a trunk group for echo cancellation

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

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

2. Go to the Trunk Features page. Note: the fields displayed depend on the trunk group type.

Trunk Features screen

```
change trunk-group n                                          Page 3 of x
TRUNK FEATURES
         ACA Assignment? _      Measured: ____
                                              Maintenance Tests? _
                        Data Restriction? _

  Abandoned Call Search? _
  Suppress # Outpulsing? _

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

3. Move to the following field

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

4. Save the changes.

TN799DP Control LAN

Systems in a private network are interconnected by both tie trunks (for voice communications) and data links (for control and transparent feature information). Various DS1, IP, and analog trunk circuit packs provide the voice-communications interface. For TCP/IP connectivity, the data-link interface is provided by a TN799DP Control LAN (C-LAN) circuit pack. (For more information about this VoIP transmission hardware, see TN799DP control LAN (C-LAN) interface on page 18 in the Network quality management section of the Networking overview chapter.)

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

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

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

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

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

Physical addressing for the C-LAN board

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

IP addressing techniques for the C-LAN board

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

Installing the TN799DP C-LAN

TCP/IP connections (Ethernet or PPP) require a TN799DP C-LAN circuit pack, unless your system has embedded Ethernet capabilities. Before you install the C-LAN circuit pack, be sure you understand the requirements of your LAN. For information about LAN requirements for VoIP, go to http://www.extremenetworks.com/LIBRARIES/Avaya/AvayaIPvoiceQualityNetworkRequirements.pdf and look in the white paper titled *Avaya IP Voice Quality Network Requirements (EF-LB1500)*.

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

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

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

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

   Note:

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

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

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

Note:

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

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

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

   (for example, **01a08** for cabinet 1, carrier A, and slot 8).

4. Enter the port bandwidths or use the defaults in the **Pt0**, **Pt1**, and **Pt2 Inter-Board Link Timeslots** fields.
5. Submit the screen.

6. Verify that the bus bridge LED is lit on the C-LAN circuit pack.

   This indicates that the packet bus is enabled.

Testing the packet bus and C-LAN circuit pack

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

To test the packet bus and C-LAN circuit pack

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

   This is for testing purposes only, and you will remove the board when finished.

2. Enter **`test pkt port-network`** ***`1 long`***

   For more information about these tests, refer to the **test pkt command** section in *Maintenance Commands for Avaya Communication Manager 2.1, Media Gateways and Servers*, 03-300191.

3. If the TN771D circuit pack was already in the cabinet, leave it there.
4. If you added the TN771D circuit pack to the cabinet in order to test the TN799DP circuit pack, remove it from the cabinet.

Installing C-LAN cables to a hub or ethernet switch

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

See Figure 26: Cable connection for C-LAN connectivity.

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

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

Figure 26: Cable connection for C-LAN connectivity

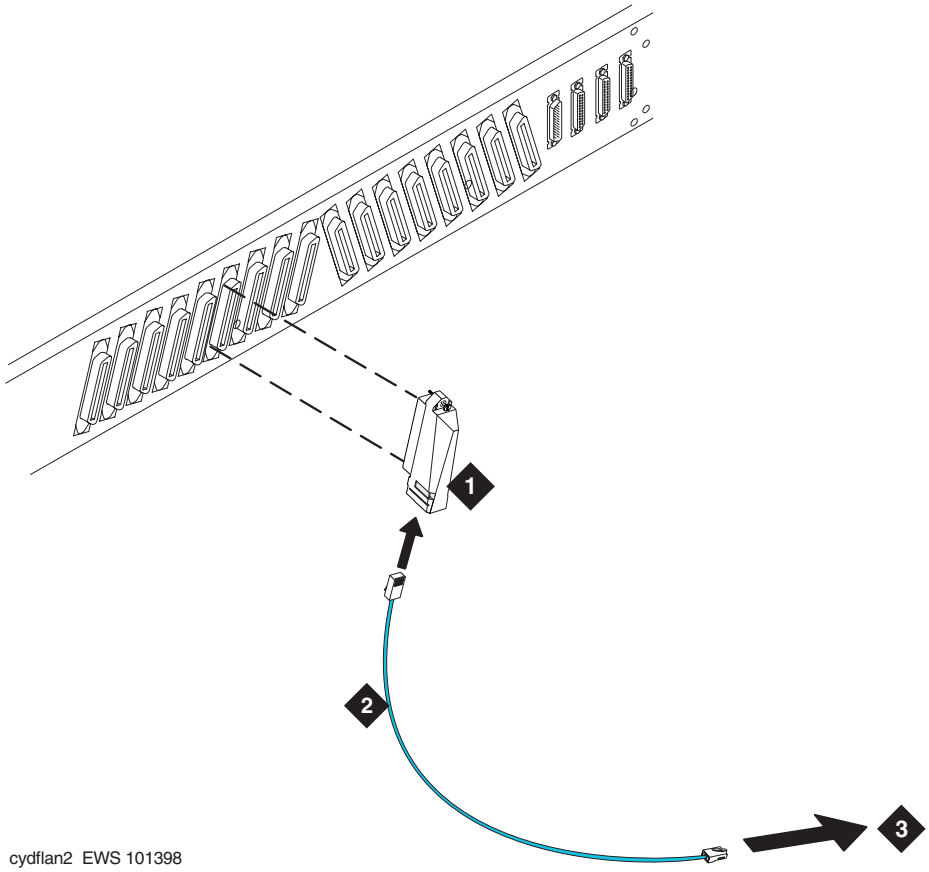


Figure notes:

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

Assigning IP node names

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

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

To assign IP node names:

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

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

    Name              IP Address
default_____________  0__.0__.0__.0__
node-1______________  192.168.10_.31_
node-2______________  192.168.10_.32_
____________________  ___.___.___.___
```

2. Enter values.

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

3. Submit the screen.

Defining a LAN default gateway

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

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

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

Setting up Alternate Gatekeeper and C-LAN load balancing

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

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

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

Note:

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

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

- IP telephone
- IP Softphone
- Avaya IP Agent

Endpoint capabilities

Table 6: Endpoint capabilities

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

Note:

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

TN2302AP IP Media Processor

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

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

The TN2302AP IP Media Processor additionally performs the functions:

- Echo cancellation
- Silence suppression

- DTMF detection
- Conferencing

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

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

Improving theTN2302AP transmission interface

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

Supporting TN2302AP hairpinning

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

Testing TN2302AP ports

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

Enabling a survivable remote EPN

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

TN2602AP IP Media Resource 320

The TN2602AP IP Media Resource 320 provides high-capacity voice over Internet protocol (VoIP) audio access to the switch for local stations and outside trunks. The IP Media Resource 320 provides audio processing for the following types of calls:

- TDM-to-IP and IP-to-TDM
- IP-to-IP

The TN2602AP IP Media Resource 320 circuit pack has two capacity options, both of which are determined by the license file installed on Communication Manager:

- 320 voice channels, considered the standard IP Media Resource 320
- 80 voice channels, considered the low-density IP Media Resource 320

Only two TN2602AP circuit packs are allowed per port network.

Note:

The TN2602AP IP Media Resource 320 is not supported in CMC1 and G600 Media Gateways.

Load balancing

Up to two TN2602AP circuit packs can be installed in a single port network for load balancing The TN2602AP circuit pack is also compatible with and can share load balancing with the TN2302 and TN802B IP Media Processor circuit packs. Actual capacity may be affected by a variety of factors, including the codec used for a call and fax support.

Note:

When two TN2602AP circuit packs, each with 320 voice channels, are used for load balancing within a port network, the total number of voice channels available is 484, because 484 is the maximum number of time slots available for a port network.

Bearer duplication

Two TN2602AP circuit packs can be installed in a single port network (PN) for duplication of the bearer network. In this configuration, one TN2602AP is an active IP media processor and one is a standby IP media processor. If the active media processor, or connections to it, fail, active connections failover to the standby media processor and remain active. This duplication prevents active calls in progress from being dropped in case of failure. The interchange between duplicated circuit packs affects only the PN in which the circuit packs reside.

Note:

The 4606, 4612, and 4624 IP telephones do not support the bearer duplication feature of the TN2602AP circuit pack. If these telephones are used while an interchange from the active to the standby media processor is in process, then calls might be dropped.

Virtual IP and MAC addresses to enable bearer duplication

Duplicated TN2602AP circuit packs in a PN share a virtual IP and virtual MAC address. These virtual addresses are owned by the currently-active TN2602. In addition to the virtual IP address, each TN2602 has a "real" IP address. All bearer packets sent to a PN that contains duplicated TN2602AP circuit packs, regardless of whether the packets originate from TN2602s

in other PNs or from IP phones or gateways, are sent to the virtual IP address of the TN2602 pair in that PN. Whichever TN2602AP circuit pack is active is the recipient of those packets.

When failover to the standby TN2602 occurs, a negotiation between TN2602s to determine which TN2602 is active and which is standby takes place. State-of-health, call state, and encryption information is shared between TN2602s during this negotiation. The newly-active TN2602AP circuit pack sends a gratuitous address resolution protocol (ARP) request to ensure that the LAN infrastructure is updated appropriately with the location of the active TN2602. Other devices within the LAN will update their old mapping in ARP cache with this new mapping.

Requirements for bearer duplication

The Communication Manager license file must have entries for each circuit pack, with the entries having identical voice channels enabled. In addition, both circuit packs must have the latest firmware that supports bearer duplication.

Duplicated TN2602AP circuit packs must be in the same subnet. In addition, the Ethernet switch or switches that the circuit packs connect to must also be in the same subnet. This shared subnet allows the Ethernet switches to use signals from the TN2602AP firmware to identify the MAC address of the active circuit pack. This identification process provides a consistent virtual interface for calls.

Combining duplication and load balancing

A single port network can up to two TN2602AP circuit packs only. As result, the port network can have either two duplicated TN2602AP circuit packs or two load balancing TN2602AP circuit packs, but not both a duplicated pair and a load-balancing pair. However, in a Communication Manager configuration, some port networks can have a duplicated pair of TN2602AP circuit packs and other port networks can have a load-balancing pair of TN2602AP circuit packs. Some port networks can also have single or no TN2602AP circuit packs.

Note:

If a pair of TN2602AP circuit packs previously used for load balancing are re-administered to be used for bearer duplication, only the voice channels of whichever circuit pack is active can be used. For example, If you have two TN2602 AP circuit packs in a load balancing configuration, each with 80 voice channels, and you re-administer the circuit packs to be in bearer duplication mode, you will have 80 (not 160) channels available. If you have two TN2602 AP circuit packs in a load balancing configuration, each with 320 voice channels, and you re-administer the circuit packs to be in bearer duplication mode, you will have 320 (rather than 484) channels available.

Features

The IP Media Resource 320 supports hairpin connections and the shuffling of calls between TDM connections and IP-to-IP direct connections. The IP Media Resource 320 can also perform the following functions:

- Echo cancellation
- Silence suppression
- Adaptive jitter buffer (320 ms)
- Dual-tone multifrequency (DTMF) detection
- AEA Version 2 and AES media encryption
- Conferencing
- QOS tagging mechanisms in layer 2 and 3 switching (Diff Serv Code Point [DSCP] and 802.1pQ layer 2 QoS)
- RSVP protocol

The TN2602AP IP Media Resource 320 circuit pack supports the following codecs for voice, conversion between codecs, and fax detection:

- G.711, A-law or Mu-law, 64 kbps
- G.726A-32 kbps
- G.729 A/AB, 8 kbps audio

The TN2602AP also supports transport of the following devices:

- Fax, Teletypewriter device (TTY), and modem calls using pass-through mode
- Fax, V.32 modem, and TTY calls using proprietary relay mode

Note:

V.32 modem relay is needed primarily for secure SCIP telephones (formerly known as Future Narrowband Digital Terminal (FNBDT) telephones) and STE BRI telephones.

- T.38 fax over the Internet, including endpoints connected to non-Avaya systems
- 64-kbps clear channel transport in support of firmware downloads, BRI secure telephones, and data appliances

Firmware download

The IP Media Resource 320 can serve as an FTP or SFTP server for firmware downloads to itself. However, this capability is activated by and available for authorized services personnel only.

As with the TN2302AP IP Media Processor, firmware upgrades of the TN2602AP circuit pack, are not call preserving. However, by using the **`campon-busyout media-processor`** command, a single or load-balanced TN2602AP circuit pack can be busied out without dropping calls, and then upgraded. In addition, with duplicated TN2602AP circuit packs, the standby TN2602AP circuit pack can be upgraded first, and then the circuit packs interchanged. The active circuit pack becomes the standby and can then be busied out and upgraded without dropping calls.

I/O adapter

The TN2602AP IP Media Resource 320 circuit pack has a services Ethernet port in the faceplate. The TN2602AP circuit pack also requires an input/output adapter that provides for one RS-232 serial port and two 10/100 Mbs Ethernet ports for LAN connections (though only the first Ethernet port is used). This Ethernet connection is made at the back of the IP Media Resource 320 slot.

Note:

The TN2302AP IP Media Processor on page 128 can also use this I/O adapter.

TN2312BP IP Server Interface (IPSI)

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

Note:

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

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

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

MM760 VoIP Media Module

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

Note:

Customers who want an essentially non-blocking system must add an additional MM760 Media Module, if they use more than two MM710 Media Modules in a single chassis. The additional MM760 provides an additional 64 channels. The MM760 is ***not*** supported on the G350 and G250 Media Gateways.

What is the MM760 Ethernet interface

The MM760 must have its own Ethernet address. The MM760 requires a 10/100 Base T Ethernet interface to support H.323 endpoints for Avaya IP trunks and stations from another G700 Media Gateway. The MM760 is not supported in the Avaya G350 Media Gateway.

Supporting voice compression on the MM760

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

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

The MM760 also supports transport of the following:

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

Note:

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

SECURITY ALERT:

Faxes sent to non-Avaya endpoints cannot be encrypted.

- Modem tones over a corporate IP intranet

Note:

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

TN8400AP Media Server circuit pack

The TN8400 Media Server circuit pack is the platform for the Avaya S8400 Media Server, which is a Linux-based server that occupies a single slot on a standard TN carrier. The S8400 Media Server efficiently provides the Avaya Communication Manager processing functions in stand-alone, single port network telephony systems requiring up to 500 stations.

For more information on the Avaya S8400 Media Server and TN8400AP Media Server circuit pack, see the section on “Linux-based media servers” in the *Hardware Description and Reference for Avaya Communication Manager* (555-245-207). For more information about administering the S8400 Media Server and TN8400 circuit pack, see *Installing and Configuring the Avaya S8400 Media Server*, 03-300678, at http://www.avaya.com/support.

TN8412AP S8400 server IP Interface

The TN8412AP S8400 server IP interface (SIPI) is used in an S8400-based system. It provides transport of control messages between the S8400 Media Server and the media server’s port network (PN) using direct connections. (Connections using the customer’s LAN and WAN are possible but not typical.) Through these control messages, the media server controls the PN.

The SIPI always resides in the tone clock slot on a media gateway and uses an Ethernet interface to connect to:

- The S8400 server
- A laptop computer connected to the server through a services port

The SIPI provides the following functions:

- PN clock generation and synchronization for Stratum 4 type II only
- PN tone generation
- PN tone detection, global call classification, and international protocols
- Environmental maintenance

The SIPI can be accessed remotely using the Telnet and SSH protocols. The SIPI can serve as an SSH client, as well, for remote access from the SIPI to the Communication Manager server. The C-LAN can also serve as an FTP or SFTP server for file transfers and firmware downloads.

Note:

The SIPI cannot serve as an SFTP client. Additionally, the SSH/SFTP capability is only for the control network interface, not the Services interface.

The SIPI supports the following functions and devices:

- Eight global call classification ports
- Network diagnostics
- Download of SIPI firmware updates using Communication Manager Web pages, the **`loadipsi`** command from the server's Linux command line, or the Software Update Manager.

The TN8412AP SIPI is compatible with the S8400 media server and the G650 gateway. It is also compatible with the G600 and CMC1 gateways in migration systems.

Note:

An S8400 system is shipped with a TN8412AP SIPI circuit pack. However, the TN2312BP IPSI circuit pack is also compatible with S8400 systems.

For more information on the S8400 Media Server and TN8412 circuit pack, see the section on "Linux-based media servers" in the *Hardware Description and Reference for Avaya Communication Manager* (555-245-207). For more information about administering the TN8412 circuit pack, see *Installing and Configuring the Avaya S8400 Media Server*, 03-300678, at http://www.avaya.com/support.

Administering Avaya gateways

The following documents additional information about the administration of the Avaya gateways:

- *Administrator Guide for Avaya Communication Manager* (03-300509).
- *Upgrading, Migrating, and Converting Media Servers and Media Gateways*, 03-300412.

Administering IP trunks

The following sections describe the administration of IP trunks:

- Administering SIP trunks
- Administering H.323 trunks

Administering SIP trunks

SIP is the Session Initiation Protocol, an endpoint-oriented messaging standard defined by the Internet Engineering Task Force (IETF). As implemented by Avaya for release 2.0 and later of Communication Manager, SIP "trunking" functionality is available on any of the Linux-based media servers (S8300, S8500 or S8700-series). These media servers function as Plain Old Telephone Service (POTS) gateways, and they also support name/number delivery between and among the various non-SIP endpoints supported by Communication Manager (analog, DCP or H.323 stations and analog, digital or IP trunks), and SIP-enabled endpoints, such as the Avaya 4600-series SIP Telephones. In addition to its calling capabilities, IP Softphone R5 and later also includes optional instant-messaging client software, which is a SIP-enabled application, while continuing its full support of the existing H.323 standard for call control. Avaya SIP Softphone R2 and later releases fully support SIP for voice call control, as well as instant messaging and presence.

For more information on SIP trunk administration and usage, see *SIP Support in Avaya Communication Manager*, 555-245-206, and for information on proxy and registrar functions on the SIP server, see *Installing and Administering SIP Enablement Services* (03-600768).

Administering H.323 trunks

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

The TN2302AP or TN2602AP enables H.323 trunk service using IP connectivity between an Avaya IP solution and another H.323 v2-compliant endpoint.

H.323 trunk groups can be configured as:

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

Setting up H.323 trunks for administration

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

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

  Note:

  These circuit packs are not required if your system has built-in Ethernet capabilities (S8300).
- Administering QoS parameters
- Assigning IP node names and IP addresses
- Defining IP interfaces (C-LAN, TN2302AP, or TN2602AP Load Balanced)
- Assigning link through Ethernet data module (S8500/S8700-series)
- Implementing Best Service Routing (optional)

Verifying customer options for H.323 trunking

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

To verify customer options for H.323 trunking:

1. Type **`display system-parameters customer-options`**, and go to the **Optional Features** screen.
2. Verify that the following fields have been completed on pages 1 and 2 of this screen:

Field	Conditions/Comments
G3 Version	This value should reflect the current version of Avaya Communication Manager.
Maximum Administered H.323 Trunks	Number of trunks purchased. Value must be greater than 0. On Page 2 of the screen.
Maximum Administered Remote Office Trunks	Number of remote office trunks purchased. This is also located on page 2 of the screen.

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

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

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

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

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

Circuit Packs screen

```
                            Page 2 of 5

                          Circuit Packs

Cabinet 1                                  Carrier: B
                                       Carrier Type: port

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

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

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

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

Fields for IP Media	Conditions/Comments
Code	**TN2302AP** or **TN2602AP**
Name	**IP Media Processor** (displays automatically)

4. Submit the screen.

Administering QoS parameters

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

Administer additional QoS parameters, including defining IP Network Regions and specifying the codec type to be used. See Chapter 4: Network quality administration.

Assigning IP node names and IP addresses

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

A node can be:

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

Enter the AUDIX name and IP address on the **AUDIX Node Names** 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 IP Node Names:

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

IP Node Names screen

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

                              IP NODE NAMES

  Name            IP Address
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       192.168.10.80
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 unique IP addresses for each node name.

3. Submit the screen.

Defining IP interfaces (C-LAN, TN2302AP, or TN2602AP Load Balanced)

The IP interface for each C-LAN, TN2302AP Media Processor, or TN2602AP (load-balanced) circuit pack on the switch must be defined on the **IP Interfaces** screen. Each switch in an IP network has one **IP Interfaces** screen.

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

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

   Note:

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

IP Interfaces screen

```
add ip-interface 01a08                                        Page 1 of x
                                  IP INTERFACES

                   Type: CLAN
                   Slot: 01A08
             Code/Suffix: TN799
               Node Name: makita-clan1
              IP Address: 172.28.5.254
             Subnet Mask: 255.255.255.0                          Link?
          Gateway Address:
   Enable Ethernet Port? y                     Allow H.323 Endpoints?
         Network Region: 20                     Allow H.248 Gateways?
                   VLAN: n                       Gatekeeper Priority?

Target socket load and Warning level: 400
      Receive Buffer TCP Window Size:

                         ETHERNET OPTIONS
              Auto? n
             Speed:100Mbps
            Duplex: Full
```

2. Complete the following fields as shown:

Field	Conditions/Comments
Critical Reliable Bearer	Appears only for the TN2602AP. Type **n** when the TN2602AP is in load balancing mode or is the only TN2602AP circuit pack in the port network.
Type	Display only. This field is automatically populated with **C-LAN, MEDPRO,** or **PROCR**. The fields differ on the screens for each of the IP Interface types. Required entries may also differ for Processor Ethernet (PE). See the Screen Reference chapter of the *Administrator Guide for Avaya Communication Manager,* 03-300509.
Slot	Display only. The slot location for the circuit pack.
Code/Suffix	Display only. This field is automatically populated with TN799DP for C-LAN, TN2302AP for IP Media Processor, or TN2602AP for IP Media Resource 320, and the suffix letter(s).
Node name	The node name for the IP interface. This node name must already be administered on the **IP Node Names** screen.
IP Address	Display only. The IP address for this IP interface. The IP address is associated with the node name on the **IP Node Names** screen.

Field	Conditions/Comments
Subnet Mask	The subnet mask associated with the IP address for this IP interface.
Link?	Display only. Shows the administered link number for an Ethernet link. See Assigning link through Ethernet data module (S8500/S8700-series) on page 148
Gateway Address	The address of a network node that serves as the default gateway for the IP interface.
Enable Ethernet Port?	Enter **y**
Allow H.323 Endpoints?	Controls whether IP endpoints can register on the interface. On a simplex main server, enter **y** to allow H.323 endpoint connectivity to the PE interface. Enter **n** if you do not want H.323 endpoint connectivity to the PE interface. **Note:** For an Enterprise Survivable Server (ESS), this field is display-only and is set to **n**. H.323 endpoint connectivity using the PE interface on an ESS server is not supported. For a Local Survivable Processor (LSP), this field is display-only and is set to **y**.
Network Region	The region number for the IP interface. Enter a value between **1-250**
Allow H.248 Gateways?	Controls whether H.28 media gateways (G700, G350, G250) can register on the interface. On a simplex main server, enter **y** to allow H.248 endpoint connectivity to the PE interface. Enter **n** if you do not want H.248 endpoint connectivity to the PE interface. **Note:** For an Enterprise Survivable Server (ESS), this field is display-only and is set to **n**. H.248 endpoint connectivity using the PE interface on an ESS server is not supported. For a Local Survivable Processor (LSP), this field is display-only and is set to **y**.
VLAN	The 802.1Q virtual LAN value (**0 - 4094**) or **n** (no VLAN). This VLAN field interfaces with the TN799 (C-LAN) or TN802B Media Processor circuit packs; it does not send any instructions to IP endpoints.
Gatekeeper Priority?	Appears only if **Allow H.323 Endpoints** is **y** and the Communication Manager server is a main server or an LSP. This field does not display on an ESS server. This field allows a priority to be set on the interface. This affects where the interface appears on the gatekeeper list. Enter the desired priority number, a value from **1** to **9**. The value in this field is used on the alternate gatekeeper list. The lower the number, the higher the priority. Default is **5**.

Field	Conditions/Comments
VOIP Channels	Appears only for a TN2602AP circuit pack. Enter the number of VoIP channels assigned to the TN2602AP circuit pack, either **0**, **80**, or **320**. **0** means the circuit pack will not be used. **Note:** If two TN2602 circuit packs in a port network are administered for 320 channels, only 512 channels are used due to the 512 TDM timeslot maximum for a port network. The system-wide number of TN2602 circuit packs administered for 80 channels cannot exceed the number of 80-channel licenses installed on system. Similarly, the number of TN2602 circuit packs administered for 320 channels cannot exceed the number of 320-channel licenses installed on the system.
Target socket load and Warning level	Always leave the default (**400**) unless instructed to enter a different value (**1** to **499**) by Avaya Services.
Receive Buffer TCP Window Size	A value of **512** to **8320**
Auto? Speed Duplex	Set Ethernet Options to match the customers network. The recommended settings are: • Auto? **n** If you set Auto to **n**, also complete the following fields. The recommended values are displayed. • Speed: **100 Mbps** • Duplex: **Full** See *IP Telephony Implementation Guide*, for a discussion of the Ethernet Options settings.

3. Submit the screen.

Defining IP interfaces (duplicated TN2602AP)

To define IP interfaces for duplicate TN2602AP Media Resource 320 circuit packs:

1. Type **`add ip-interface`** ***`CCcccss`*** to open the **IP Interfaces** screen.

   The IP Interfaces screen appears.

Note:

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

```
add ip-interface 1a03                                        Page 1 of 1
                                  IP INTERFACES

                            Critical Reliable Bearer? n
                    Type: MEDPRO
                    Slot: 01A03
              Code/Suffix: TN2602
                Node Name: medres03a01
              IP Address: 192.168.1.82
              Subnet Mask: 255.255.255.0
          Gateway Address: . . .
     Enable Ethernet Port? y
           Network Region: 1
                    VLAN: n



                                  ETHERNET OPTIONS
                    Auto? n
                   Speed: 100 Mbps
                  Duplex: Full
```

2. In the **Critical Reliable Bearer?** field, type **y**, and press **Enter**.

   A second column of data for a standby TN2602AP appears on the right of the screen.

```
add ip-interface 1a03                                        Page 1 of 1
                                  IP INTERFACES

                            Critical Reliable Bearer? y
                    Type: MEDPRO
                    Slot: 01A03                          Slot:
              Code/Suffix: TN2602                  Code/Suffix:
                Node Name: medpro03a01               Node Name:
              IP Address: 192.168.1.82             IP Address:
              Subnet Mask: 255.255.255.0
          Gateway Address: . . .
     Enable Ethernet Port? y              Enable Ethernet Port? y
           Network Region: 1
                    VLAN: n                              VLAN: n
            VOIP Channels: xxx
 Shared Virtual Address: 255.255.255.255
       Virtual MAC Table:                  Virtual MAC Address:
                                  ETHERNET OPTIONS
                    Auto? n                              Auto? n
                   Speed: 100 Mbps                       Speed: 100 Mbps
                  Duplex: Full                         Duplex: Full
```

3. Complete the following fields as shown:

Field	Conditions/Comments
Type	Display only. This field is automatically populated with **MEDPRO**.
Slot	Slot location entered in the command line. Enter the location of the second TN2602AP circuit pack for a non-duplicated board. The second (right-side) Slot field is automatically populated when Critical Reliable Bearer is **y**.
Code/Sfx	Circuit pack TN code and suffix. Display only for TN2602AP when Critical Reliable Bearer is **n**. The second (right-side) Code/Sfx field is automatically populated based on the corresponding Slot field information, when Critical Reliable Bearer is **y**.
Node name	The node name for the IP interface. This node name must already be administered on the **IP Node Names** screen.
IP Address	Display only. The IP address for this IP interface. The IP address is associated with the node name on the **IP Node Names** screen.
Subnet Mask	Enter the Subnet Mask for TN2602AP. This entry also applies to the second TN2602AP circuit pack when Critical Reliable Bearer is **y**
Gateway Address	The IP address of the LAN gateway associated with the TN2602AP. This entry also applies to the second TN2602AP circuit pack when Critical Reliable Bearer is **y**
Enable Ethernet Pt	**y**/**n** **y** = The Ethernet Port associated with the TN2602AP is in service. If this is an active board, set to **n** only when there is no standby, or when the standby has been disabled. **Note:** Note: You may be required to enter **n** in this field before you make changes to this screen.
Network Region	Number of the Network Region where the interface resides. This entry also applies to the second TN2602AP circuit pack when Critical Reliable Bearer is **y**
VLAN	The 802.1Q virtual LAN value (**0 - 4094**) or **n** (no VLAN). This VLAN field interfaces with the media processor circuit packs; it does not send any instructions to IP endpoints.
	1 of 3

Field	Conditions/Comments
VOIP Channels	**0** (will not support voice calls) **80** (low density) **320** (standard) The number of VoIP channels that are allocated to the associated TN2602. Appears for a TN2602 circuit pack on Communication Manager 3.0/V13 or greater. This number also applies to the second TN2602AP circuit pack when Critical Reliable Bearer is **y** Users will be blocked from administering 80 or 320 VoIP channels if there is no available capacity for the corresponding "Maximum TN2602 boards with 80 VoIP Channels"/"Maximum TN2602 boards with 320 VoIP Channels" license feature.
Shared Virtual Address	The virtual IP address shared by the two TN2602AP circuit packs, when duplicated. This address enables Communication Manager to connect endpoints through the TN2602AP circuit packs to the same address, regardless of which one is actually active. Appears when Critical Reliable Bearer is **y**.
Virtual MAC Table	**1** through **4**, default = **1** Table number where the virtual MAC address, shared by duplicated TN2602AP circuit packs, is obtained. Appears when Critical Reliable Bearer is **y**. You might choose a different table number other than **1** if all of the following conditions exist: • A port network under the control of a different Communication Manager main server has duplicated TN2602AP circuit packs. • That port network controlled by a different main server has the same number as the port network in which you are administering the TN2602AP circuit packs. • The port network or its main server connects to the same Ethernet switch as the port network in which you are administering the TN2602AP circuit packs. Selecting a different Virtual MAC Table from that chosen for a port network that has the previously-listed conditions helps prevent the possibility that two TN2602AP circuit packs within the customer's network will have the same virtual MAC address.

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Field	Conditions/Comments
Virtual MAC Address	Virtual MAC address that is shared by duplicated TN2602AP circuit packs. Automatically populated based on the Virtual MAC address table. Appears when Critical Reliable Bearer is **y**.
Auto?	Set Ethernet Options to match the customers network. The recommended settings are: • Auto? **n** If you set Auto to **n**, also complete the following fields. The recommended values are displayed. • Speed: **100 Mbps** • Duplex: **Full** See *IP Telephony Implementation Guide*, for a discussion of the Ethernet Options settings.
	3 of 3

4. Submit the screen.

Assigning link through Ethernet data module (S8500/S8700-series)

Note:

The S8300 Media Server does not support data modules.

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

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

To assign a link through an Ethernet data module:

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

Data Module screen

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

   Data Extension: 700              Name:__________
             Type: Ethernet
             Port:
             Link:


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

2. Complete the following fields as shown:

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

For more information on the fields that may appear on this screen, see the *Administrator Guide for Avaya Communication Manager,* 03-300509.

3. Submit the screen.

Implementing Best Service Routing (optional)

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

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

Administering H.323 trunks

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

- Creating an H323 trunk signaling group

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

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

Creating an H323 trunk signaling group

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

Note:

The following steps address only those fields that are specifically related to H.323 trunks. The other fields are described in the *Administrator Guide for Avaya Communication Manager,* 03-300509.

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

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

Signaling Group screen

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

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

Near-end Node Name: clan-a1          Far-end Node Name: clan-b1
Near-end Listen Port: 1720          Far-end Listen Port: 1720
                                 Far-end Network Region:
          LRQ Required? n                 Calls Share IP Signaling Connection? n
          RRQ Required? n
      Media Encryption? y
            Passphrase:                       Bypass If IP Threshold Exceeded? y
                                                     H.235 Annex H Required? n
            DTMF over IP: out-of-band         Direct IP-IP Audio Connections? y
  Link Loss Delay Timer(sec): 90                       IP Audio Hairpinning? n
         Enable Layer 3 Test? y                  Interworking Message: PROGress
                                               DCP/Analog Bearer Capability: 3.1kHz
```

2. Complete the following fields as shown:

Table 7: 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.
T303 Timer	Use this field to enter the number of seconds the system waits for a response from the far end before invoking Look Ahead Routing. Appears when the Group Type field is isdn-pri (DS1 Circuit Pack screen) or h.323 (Signaling Group screen).
Near-end Node Name	Enter the node name for the C-LAN IP interface on this switch. The node name must be administered on the **Node Names** screen and the **IP Interfaces** screen.

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Table 7: Signaling Group screen options (continued)

Field	Conditions/Comments
Far-end Node Name	This is the node name for the far-end C-LAN IP Interface used for trunks assigned to this signaling group. The node name must be administered on the **Node Names** screen on this switch. Leave blank when the signaling group is associated with an unspecified destination.
Near-end Listen Port	Enter an unused port number from the range **1719**, **1720** or **5000–9999**. Avaya recommends **1720**. If the **LRQ** field is **y**, enter **1719**.
Far-end Listen Port	Enter the same number as the one in the **Near-end Listen Port** field. This number must match the number entered in the **Near-end Listen Port** field on the signaling group screen for the far-end switch. Leave blank when the signaling group is associated with an unspecified destination.
Far-end Network Region	Identify network assigned to the far end of the trunk group. The region is used to obtain the codec set used for negotiation of trunk bearer capability. If specified, this region is used instead of the default region (obtained from the C-LAN used by the signaling group) for selection of a codec. Enter a value between **1-250**. Leave blank to select the region of the near-end node (C-LAN).
LRQ Required	Enter **n** when the far-end switch is an Avaya product and H.235 Annex H Required? is set to **n**. Enter **y** when: ● H.235 Annex H Required? is set to **y**, or ● the far-end switch requires a location request to obtain a signaling address in its signaling protocol.
Calls Share IP Signaling Connection	Enter **y** for connections between Avaya equipment. Enter **n** when the local and/or remote switch is not Avaya's.
RRQ Required	Enter **y** when a vendor registration request is required.
Bypass if IP Threshold Exceeded?	Enter **y** to automatically remove from service trunks assigned to this signaling group when IP transport performance falls below limits administered on the **Maintenance-Related System Parameters** screen.

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Table 7: Signaling Group screen options (continued)

Field	Conditions/Comments
H.235 Annex H Required?	Enter **y** to indicate that the CM server requires the use of H.235 amendment 1 with annex H protocol for authentication during registration.
DTMF Over IP?	SIP trunks only. Support for SIP Enablement Services (SES) trunks requires the default entry of **rtp-payload**.
Direct IP-IP Audio Connections?	Allows direct audio connections between H.323 endpoints. For SIP Enablement Services (SES) trunk groups, this is the value that allows direct audio connections between SES endpoints. Enter a **y** to save on bandwidth resources and improve sound quality of voice over IP (VoIP) transmissions.
IP Audio Hairpinning?	The **IP Audio Hairpinning** field entry allows the option for H.323 and SIP Enablement Services (SES)-enabled endpoints to be connected through the IP circuit pack in the media server or switch, without going through the time division multiplexing (TDM) bus. Type **y** to enable hairpinning for H.323 or SIP trunk groups. Default is **n**.
Interworking Message	This field determines what message Avaya Communication Manager sends when an incoming ISDN trunk call interworks (is routed over a non-ISDN trunk group). Normally select the value, **PROGress**, which asks the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk. Selecting the value **ALERTing** causes the public network in many countries to play ringback tone to the caller. Select this value only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.
DCP/Analog Bearer Capability	This field sets the information transfer capability in a bearer capability IE of a setup message to **speech** or **3.1kHz**. The latter is the default. The default value provides 3.1kHz audio encoding in the information transfer capability. Selecting the value of **speech** provides speech encoding in the information transfer capability.

3 of 3

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

   Enter NCA TSC information on this screen according the detailed descriptions contained in the Screen Reference chapter of the *Administrator Guide for Avaya Communication Manager,* 03-300509.

4. Submit the screen.

Creating a trunk group for H.323 trunks

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

Note:

The following steps address only those fields that are specifically related to H.323 trunks. The other fields are described in the *Administrator Guide for Avaya Communication Manager,* 03-300509.

To create an ISDN trunk group, do the following:

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

Trunk Group screen

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

 Group Number: 3__                   Group Type: isdn          CDR Reports: y
   Group Name: TG 3 for H.323 trunks       COR: 1       TN: 1__       TAC: 103
    Direction: two-way        Outgoing Display? n          Carrier Medium: H.323
  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? unre
```

```
add trunk-group next                                          Page 2 of x
                                TRUNK GROUP

      Group Type: isdn
TRUNK PARAMETERS
       Codeset to Send Display: 0     Codeset to Send National IEs: 6
      Max Message Size to Send: 260                  Charge Advice: none
Supplementary Service Protocol: a     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: cyclical                        QSIG Value-Added? n
                                                      Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                   Format:
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
          Administer Timers? n
```

2. Complete the following fields as shown:

Field	Conditions/Comments
Group Type	Enter **isdn**
Carrier Medium	Enter **H.323**
Service Type	Enter **tie**
TestCall ITC	Enter **unre** (unrestricted).
TestCall BCC	Enter **0**
Codeset to Send Display	Enter **0**
Outgoing Display	This field might need to be changed if the far-end is not Avaya's.

3. Go to the **Trunk Features** page of this screen.

Trunk Features screen

```
add trunk-group next                                          Page   3 of  x
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                                          Maintenance Tests? y
                                Data Restriction? n     NCA-TSC Trunk Member:
                                       Send Name: y      Send Calling Number: y
            Used for DCS? n                             Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: public
 Outgoing Channel ID Encoding: exclusive     UUI IE Treatment: service-provider

                                                  Replace Restricted Numbers? n
                                                 Replace Unavailable Numbers? n
                                                       Send Connected Number: n
                                                  Hold/Unhold Notifications? n
               Send UUI IE? y
                 Send UCID? n
    Send Codeset 6/7 LAI IE? y                          DS1 Echo Cancellation? n

       Apply Local Ringback? n            US NI Delayed Calling Name Update? n
Show ANSWERED BY on Display? y
                            Network (Japan) Needs Connect Before Disconnect? n
```

4. Complete the following fields as shown:

Field	Conditions/Comments
Send Name Send Calling Number Send Connected Number	If **y** is entered, either the **ISDN Numbering - Public/Unknown Format** screen, or the **ISDN Numbering - Private** screen (based on the **Format** field) is accessed to construct the actual number to be sent to the far end.

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

```
add trunk-group next                                        Page 6 of  x
                                  TRUNK GROUP
                                            Administered Members (min/max):   0/0
GROUP MEMBER ASSIGNMENTS                        Total Administered Members:   0

       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
  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 the trunks to the existing or 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 for the signaling group associated with this H.323 trunk.

Modifying the H.323 trunk signaling group

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

To modify an H.323 trunk signaling group:

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

Signaling Group screen

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

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

Near-end Node Name: clan-a1          Far-end Node Name: clan-b1
Near-end Listen Port: 1720          Far-end Listen Port: 1720
                                  Far-end Network Region:
          LRQ Required? n                 Calls Share IP Signaling Connection? n
          RRQ Required? n
      Media Encryption? y
            Passphrase:                       Bypass If IP Threshold Exceeded? y
                                                     H.235 Annex H Required? n
            DTMF over IP: out-of-band          Direct IP-IP Audio Connections? y
  Link Loss Delay Timer(sec): 90                         IP Audio Hairpinning? n
         Enable Layer 3 Test? y                    Interworking Message: PROGress
                                               DCP/Analog Bearer Capability: 3.1kHz
```

3. Complete the following field:

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

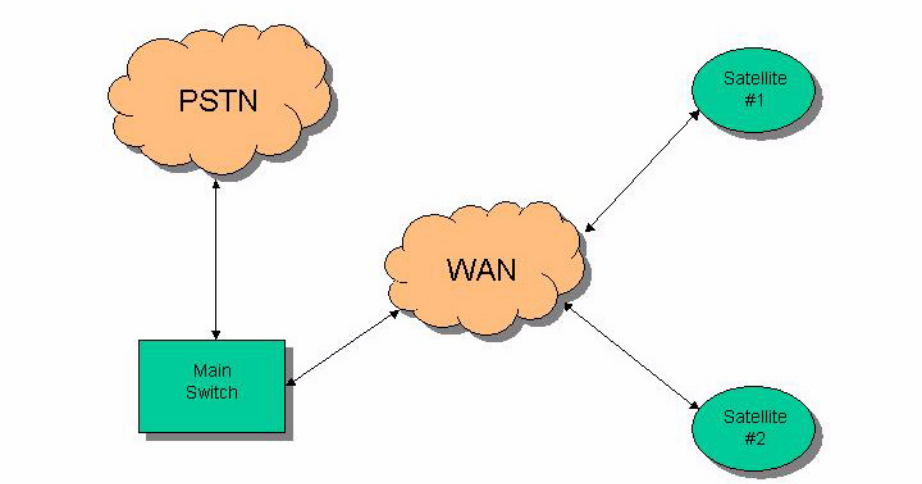
4. Submit the screen.
5. Type **release signaling-group *number*** to release the signaling group.

Dynamic generation of private/public calling party numbers

Often it is necessary to generate a private Calling Party Number (CPN) for calls within a network, but a public CPN for calls that route through the main network switch to the PSTN.

Consider a network such as the following:

Private/public calling party numbers (CPN)



In this network, the customer wants to use internal numbering among the nodes of the network (for example, a 4-digit Uniform Dial Plan (UDP)), but when any node dials the PSTN, to route the call to the PSTN through the main switch.

On page 2 of the ISDN **Trunk Group** screen, set the **Numbering Format** field to **private** or **unk-pvt**. (The value **unk-pvt** means "encode the number as an "unknown" type of number, but use the **Numbering-Private Format** screen to generate the actual number.)

Note:

IP trunks function as ISDN trunks in this respect.

In the network example, the system only generates a Private CPN if the caller dials a Private (level 0/1/2) or Unknown (unk-unk) number. If the caller dials a Public number, the system generates a Public CPN. It is necessary to fill out the **Numbering-Private Format** and **Numbering-Public/Unknown Format** forms appropriately, and then to set the IP trunk groups on the two satellites to use **private** or **unk-pvt Numbering Format** for their CPNs.

Note:

You can designate the type of number for an outgoing call as Private (level 0/1/2) either on the **AAR Analysis** screen or on the **Route Pattern** screen, but you can only designate the type of number as Unknown (**unk-unk**) on the **Route Pattern** screen. If the customer uses UDP, Unknown is the better Type of Number to use.

The default **Call Type** on the **AAR Analysis** screen is **aar**. For historical reasons, **aar** maps to a "public" numbering format. Therefore, you must change the **Call Type** for calls within your network from **aar** to a **private** or **unk-unk** type of number. For a UDP environment, the recommended way is to set the **Numbering Format** to **unk-unk** on the **Route Pattern** screen.

Administering Avaya phones

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

- Administering IP Softphones
- Installing and administering Avaya IP telephones

Administering IP Softphones

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

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

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

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

Documentation on how to set up and use the IP Softphones is included on the CD-ROM containing the IP Softphone software. Procedures for administering Communication Manager to support IP Softphones are given in *Administrator Guide for Avaya Communication Manager,* 03-300509*.*

Administering the IP Softphone

This section focuses on administration for the trunk side of the Avaya IP Solutions offer, plus a brief checklist of IP Softphone administration. Comprehensive information on the administration of IP Softphones is given in *Administrator Guide for Avaya Communication Manager,* 03-300509.

There are two main types of IP Softphone configurations:

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

Communication Manager can distinguish between various IP stations at RAS using the product ID and release number sent during registration. An IP phone with an Avaya manufacturer ID can register if the number of stations with the same product ID and the same or lower release number *is less than* the administered system capacity limits. System limits are based on the number of simultaneous registrations. Note that a license is required for each station that is to be IP softphone enabled.

Administering a Telecommuter phone

The Telecommuter uses two connections: one to the PC over the IP network and another connection to the telephone over the PSTN. IP Softphone PC software handles the call signaling. With IP Softphone R5 or greater, iClarity is automatically installed to handle voice communications.

Note:

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

To administer a Telecommuter phone:

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

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

Field	Value
Maximum Concurrently Registered IP Stations	Identifies the maximum number of IP stations that are simultaneously registered, not the maximum number that are simultaneously administered. This value must be greater than **0**, and must be less than or equal to the value for Maximum Ports.
Maximum Concurrently Registered Remote Office Stations	Specifies the maximum number of remote office stations that are simultaneously registered, not the maximum number that are simultaneously administered. This value must be greater than **0**, and must be less than or equal to the value for Maximum Ports.
IP Stations	This value should be **y**.

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

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

3. Go to page 2; verify whether the field **Service Link Mode:** *`as-needed`* is set as shown.
4. Install the IP Softphone software on the user's PC.

Administering a Road-warrior phone

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

Note:

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

To administer a Road-warrior phone:

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

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

Field	Value
Maximum Concurrently Registered IP Stations	Specifies the maximum number of IP stations that are simultaneously registered, not the maximum number that are simultaneously administered. This value must be greater than **0**.
IP Stations	Must be **y**.
Product ID	This is a 10-character field that allows any character string. For new installations, IP Soft, IP Phone, IP Agent and IP ROMax product IDs automatically display.
Rel. (Release)	Identifies the release number
Limit	Defaults to **1**

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

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

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

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

Installing and administering Avaya IP telephones

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

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

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

About the 4600-series IP telephones

The 4600-series IP Telephone product line possesses a number of shared model features and capabilities. All models also feature

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

The 4600-series IP Telephone product line includes the following telephones:

- Avaya 4601 IP telephone
- Avaya 4602SW IP telephone
- Avaya 4606 IP telephone
- Avaya 4610SW IP telephone
- Avaya 4620SW/4621SW IP telephone
- Avaya 4622SW IP telephone
- Avaya 4624 IP telephone
- Avaya 4625SW IP telephone
- Avaya 4630SW IP Screenphone
- Avaya 4690 IP conference telephone

Support for SIP-enabled applications may be added to several of these IP telephones via a model-specific firmware update. See the Avaya Firmware Download Web site for more details.

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

About IP telephone hardware/software requirements

Note:

Communication Manager requires that IP telephones still running R2.1 or earlier software be upgraded to R2.2.1 or newer software. Earlier software used a dual connection architecture that is no longer supported.

4600-series IP Telephones are shipped from the factory with operational firmware installed. Some system-specific software applications are downloaded from a TFTP server through automatic power-up or reset. The 4600-series IP Telephones search and download new firmware from the TFTP server before attempting to register with Communication Manager.

During a Communication Manager upgrade, any data in the /tftpboot directory is overwritten with new software and firmware. For more detailed information on managing the firmware and configuration files for the 4600-series IP telephones during Communication Manager upgrades, see *Installing and Upgrading the Avaya G700 Media Gateway and Avaya S8300 Media Server* (555-234-100), or *Upgrading, Migrating, and Converting Media Servers and Gateways* (03-300412).

The software treats the 4600-series IP Telephones as any new station type, including the capability to **`list/display/change/duplicate/alias/remove station`**.

Note:

Audio capability for the IP Telephones requires the presence of the TN2302AP IP Media Processor or TN2602AP Media Resource 320 circuit pack, either of which provide hairpinning and IP-IP direct connections. Using a media processor resource conserves TDM bus and timeslot resources and improves voice quality.

The 4600-series IP Telephone also requires a TN799DP Control- LAN (C-LAN) circuit pack for the signaling capability on the DEFINITY Server csi platform. You do not need a C-LAN circuit pack to connect an IP Telephone if your system has built-in (for example, using an Avaya S8300 Media Server or Avaya S8700-series Media Server) or Processor Ethernet capability.

To install required TN2302AP, TN2602AP, and TN799DP circuit packs, if necessary

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

Note:

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

Administering Avaya IP telephones

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

To add an IP telephone

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

Station screen

```
add station next                                              Page 1 of 5
                                STATION

       Extension: 2010           Lock Messages? n                   BCC: 0
            Type: 4624           Security Code:                      TN: 1
            Port: IP           Coverage Path 1:                     COR: 1
            Name:              Coverage Path 2:                     COS: 1
                               Hunt-to Station:
STATION OPTIONS
                                                  Time of Day Lock Table:
              Loss Group: 2                Personalized Ringing Pattern: 1
                                                     Message Lamp Ext: 2010
           Speakerphone: 2-way                    Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal                  Media Complex Ext:
   Survivable Trunk Dest? y                            IP Softphone? y
```

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

Field	Value
Extension Type	Enter the IP Telephone 4600-series model number, such as **4624**. The following phones are administered with an alias: • 4601 (administer as a 4602) • 4602SW (administer as a 4602) • 4690 (administer as a 4620)
Port	Enter **x**, or **IP**.

Note:

A 4600-series IP Telephone is always administered as an X port, and then once it is successfully registered by the system, a virtual port number will be assigned. (Note that a station that is registered as “unnamed” is not associated with any logical extension or administered station record.)

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

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

4. Submit the screen.

About hairpinning and shuffling

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

Shuffling and hairpinning techniques differ in the way that they bypass the unnecessary call-path resources (compare either Figure 27: Shuffled audio connection between IP endpoints in the same network region on page 168 or Figure 28: Shuffled audio connection between IP endpoints in different network regions on page 169 with Figure 29: Hairpinned audio connection between 2 IP endpoints in the same network region on page 172).

Shuffled or hairpinned connections:

- Conserve channels on the TN2302AP IP Media Processor and TN2602AP IP Media Resource 320.
- Bypass the TDM bus, conserving timeslots.
- Improve voice quality by bypassing the codec on the TN2302AP IP Media Processor and TN2602AP IP Media Resource 320 circuit packs.

Because shuffling frees up more resources on the TN2302AP IP Media Processor and TN2602AP IP Media Resource 320 circuit packs than hairpinning does, Communication Manager first checks both endpoints to determine whether the Determining if shuffling is possible on page 167 are met. If the shuffling criteria are not met, Communication Manager routes the call according to the What are the criteria for hairpinning on page 171, if hairpinning is enabled. If hairpinning is not enabled, Communication Manager routes the call to the TDM bus. Both endpoints must connect through the same TN2302AP IP Media Processor and TN2602AP IP Media Resource 320 for Communication Manager to shuffle or hairpin the audio connection.

For information on interdependencies that enable hairpinning and shuffling audio connections, see Hairpinning and shuffling administration interdependencies on page 173. For a discussion of Network Address Translation (NAT), see About Network Address Translation (NAT) on page 174.

What hardware and endpoints are required

The TN2302AP IP Media Processor or TN2602AP IP Media Resource 320 circuit pack is required for shuffling or hairpinning audio connections.

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

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

What are shuffled audio connections

Shuffling an audio connection between two IP endpoints means rerouting voice channel away from the usual TDM bus connection and creating a direct IP-to-IP connection. Shuffling saves such resources as TN2302AP or TN2602AP channels and TDM bus time slots and improves voice quality because the shuffled connection bypasses the TN2302AP's or TN2602AP's codec. Both endpoints must be capable of shuffling (support H.245 protocol) before Communication Manager can shuffle a call.

Determining if shuffling is possible

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

- A point-to-point voice connection exists between two endpoints.
- No other active call (in-use or held) that requires TDM connectivity (for example, applying tones, announcement, conferencing, and others) exists on either endpoint.
- The endpoints are in the same network region or in different, interconnected regions.
- Both endpoints or connection segments are administered for shuffling by setting the **Direct IP-IP Audio Connections** field on the Station screen on page 185 or the Signaling group screen on page 183) to **y**.
- If the **Direct IP-IP Audio Connections** field is **y** (yes), but during registration the endpoint indicates that it does not support audio shuffling, then a call cannot be shuffled.

  If the **Direct IP-IP Audio Connections** field is **n** (no), but during registration the endpoint indicates that it can support audio shuffling, then calls to that endpoint cannot be shuffled, giving precedence to the endpoint administration.
- The rules for Inter-network region connection management on page 180 are met.
- There is at least one common codec between the endpoints involved and the Inter-network region Connection Management codec list.
- The endpoints have at least one codec in common as shown in their current codec negotiations between the endpoint and the switch.
- Both endpoints can connect through the same TN2302AP IP Media Processor or TN2602AP IP Media Resource 320 circuit packs.

What are shuffling examples

Shuffling within the same network region

Figure 27: Shuffled audio connection between IP endpoints in the same network region on page 168 and Figure 28: Shuffled audio connection between IP endpoints in different network regions on page 169 provide examples of shuffled audio connections.

Figure 27: Shuffled audio connection between IP endpoints in the same network region

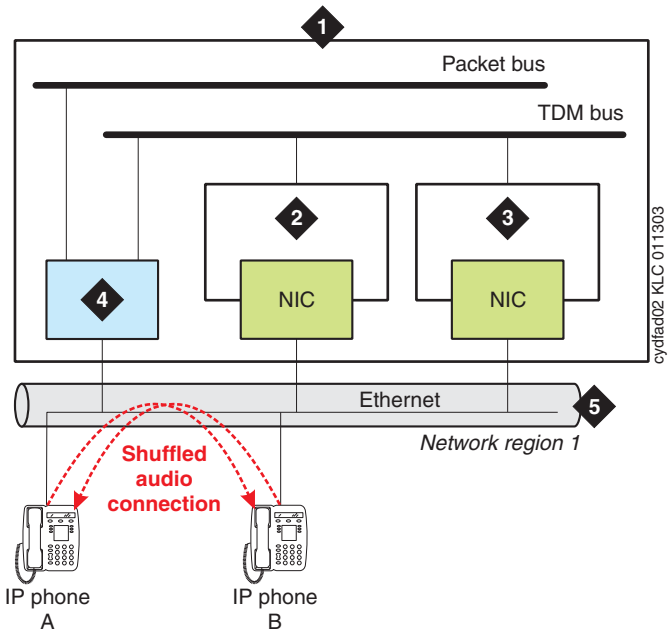


Figure notes:

1. **Avaya server**
2. **TN2302AP IP Media Processor and TN2602AP IP Media Resource 320 circuit pack**
3. **TN2302AP IP Media Processor and TN2602AP IP Media Resource 320 circuit pack**
4. **TN799 Control LAN (C-LAN) circuit pack**
5. **LAN/WAN segment administered in Communication Manager as network region 1.**

Figure 27: Shuffled audio connection between IP endpoints in the same network region on page 168 is a schematic of a shuffled connection between two IP endpoints within the same network region. After the call is shuffled, the IP Media Processors are out of the audio connection, and those channels are free to serve other media connections.

Shuffling between different network regions

Figure 28: Shuffled audio connection between IP endpoints in different network regions

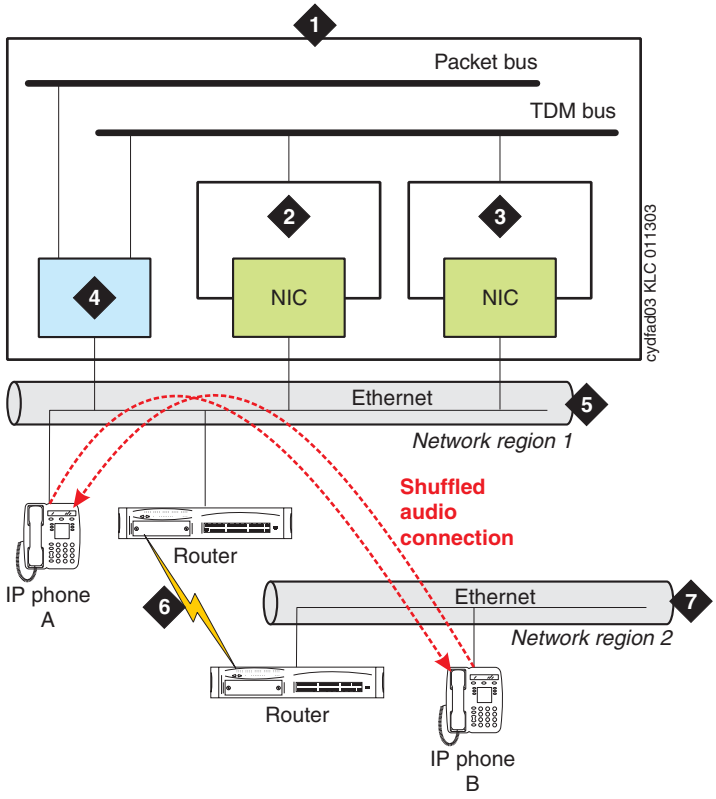


Figure notes:

1. **Avaya server**
2. **TN2302AP IP Media Processor and TN2602AP IP Media Resource 320 circuit pack**
3. **TN2302AP IP Media Processor and TN2602AP IP Media Resource 320 circuit pack**
4. **TN799 Control LAN (C-LAN) circuit pack**
5. **LAN/WAN segment administered in Communication Manager as network region 1.**
6. **IP voice packet path between LAN routers**
7. **LAN/WAN segment administered in Communication Manager as network region 2.**

Figure 28: Shuffled audio connection between IP endpoints in different network regions on page 169 is a schematic of a shuffled audio connection between two IP endpoints that are in different network regions that are interconnected and the inter-network region connection management rules are met. After the call is shuffled, both Media Processors are bypassed, making those resources available to serve other media connections. The voice packets from IP endpoints flow directly between LAN routers.

Determining whether an endpoint supports shuffling

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

To determine whether an endpoint supports shuffling:

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

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

   In the **AUDIO CHANNEL** section note the value of the **Audio** field under the **Switch Port** column.
   - If the values are the same, the endpoint is capable of shuffling.

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

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

Administrable loss plan

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

What are hairpinned audio connections

Hairpinning means rerouting the voice channel connecting two IP endpoints so that the voice channel goes through the TN2302AP IP Media Processor and TN2602AP IP Media Resource 320 circuit packs in IP format instead of through the TDM bus. Communication Manager provides only shallow hairpinning, meaning that only the IP and Real Time Protocol (RTP) packet headers are changed as the voice packets go through the TN2302AP or TN2602AP circuit pack. This requires that both endpoints use the same codec (coder/decoder), a circuit that takes a varying-voltage analog signal through a digital conversion algorithm to its digital equivalent or vice-versa (digital to analog). Throughout this section, when the word "hairpin" is used, it means shallow hairpinning.

What are the criteria for hairpinning

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

- A point-to-point voice connection exists between two endpoints.
- The endpoints are in the same network region, or in different, interconnected regions.
- A single TN2302AP IP Media Processor or TN2602AP IP Media Resource 320 circuit pack serves both endpoints.
- The endpoints use a single, common codec.
- The endpoints are administered for hairpinning: the **Direct IP-IP Audio Connections** field on the Station screen on page 185 or the Signaling group screen on page 183) is **y**.
- If the **IP Audio Hairpinning** field is **y** (yes), but during registration the endpoint indicates that it does not hairpinning, then a call cannot be hairpinned.

  If the **IP Audio Hairpinning** field is **n** (no), but during registration the endpoint indicates that it can support hairpinning, then calls to that endpoint cannot be hairpinned, giving precedence to the endpoint administration.
- The Determining if shuffling is possible on page 167 are *not* met.
- Both endpoints can connect through the same TN2302AP IP Media Processor or TN2602AP IP Media Resource 320 circuit pack.

What is an example of a hairpinned call

Hairpinned audio connections:

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

Figure 29: Hairpinned audio connection between 2 IP endpoints in the same network region on page 172 is a schematic of a hairpinned audio connection between two IP endpoints in the same network region.

Figure 29: Hairpinned audio connection between 2 IP endpoints in the same network region

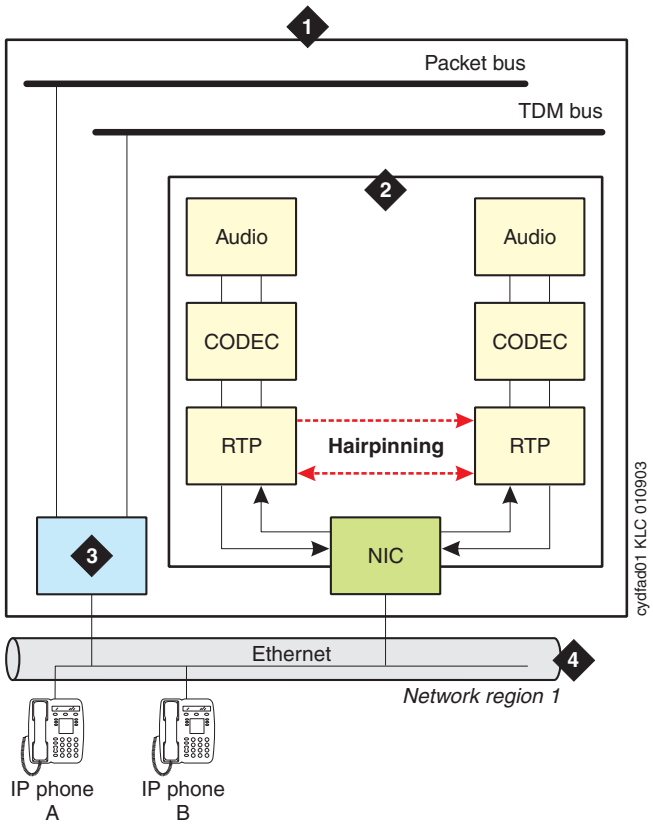


Figure notes:

1. **Avaya server**
2. **TN2302AP IP Media Processor and TN2602AP IP Media Resource 320 circuit pack**
3. **TN799 Control LAN (C-LAN) circuit pack**
4. **LAN/WAN segment administered in Communication Manager as network region 1.**

Figure 29: Hairpinned audio connection between 2 IP endpoints in the same network region on page 172 shows that hairpinned calls bypass the TN2302AP's or TN2602AP's codec, thus freeing those resources for other calls. The necessary analog/digital conversions occur in the common codec in each endpoint.

What causes a hairpinned call to be redirected

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

Determining which TN2302AP or TN2602AP circuit pack is hairpinning

Whenever a TN2302AP IP Media Processor or TN2602AP IP Media Resource 320 circuit pack is hairpinning any calls, its yellow LED is on steady. Although there is no simple way to identify all of the extension numbers that are hairpinning through a particular TN2302AP or TN2602AP circuit pack, you can determine which TN2302AP or TN2602AP circuit pack a particular extension is using for hairpinning.

To determine which TN2302AP or TN2602AP circuit pack is hairpinning:

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

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

Hairpinning and shuffling administration interdependencies

Table 8: Hairpinning and shuffling administration on page 174 summarizes the Communication Manager interdependencies that enable hairpinning and shuffling audio connections.

Note:

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

Important:

Encryption must be *disabled* for hairpinning to work, because encryption requires the involvement of resources that are not used in the shallow hairpinning connection. This not the case for shuffling, however.

Table 8: Hairpinning and shuffling administration

Administration screen	Required customer options[1]	Other interactions
Station	IP Stations Remote Office	Hairpinning is not available if **Service Link Mode** field on ***Station*** screen is **permanent**. Shuffling is available only for these endpoints[2]: • Avaya IP telephone R2 • Avaya IP Softphone (R2 or older)
Signaling group	H.323 Trunks	
Inter network region	H.323 Trunks IP Stations Remote Office	User login must have features permissions.
Feature-Related System Parameters	H.323 Trunks IP Stations Remote Office	

1. The fields listed in this column must be enabled through the License File. To determine if these customer options are enabled, use the **`display system-parameters customer-options`** command. If any of the fields listed in this column are not enabled, then either the fields for hairpinning and shuffling are not displayed or, in the case of the **Inter Network Region Connection Management** screen, the second page (the actual region-to-region connection administration) does not display.
2. Although other vendors' fully H.323v2-compliant products should have shuffling capability, you should test that before administering such endpoints for hairpinning or shuffling. See the section titled Determining whether an endpoint supports shuffling on page 170.

About Network Address Translation (NAT)

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

Note:

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

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

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

What are the types of NAT

Static 1-to-1 NAT

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

Dynamic Many-to-1 NAT

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

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

Dynamic Many-to-a-Pool NAT

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

What are the issues between NAT and H.323

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

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

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

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

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

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

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

Avaya Communication Manager NAT Shuffling feature

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

Terms:

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

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

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

  It could be a portal to the gatekeeper, such as a C-LAN, or the gatekeeper itself, such as an S8300 Media Server.

- Gateway — The Avaya device that is handling media conversion between TDM and IP, such as a MedPro board, G700 VoIP Media Module, or G350 Media Gateway.

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

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

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

Rules:

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

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

Administering hairpinning and shuffling

Choosing how to administer hairpinning and shuffling

You can administer shuffled and hairpinned connections:

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

Table 9: Hairpinning and shuffling administration on page 178 lists the forms and provides links to all three levels:

Table 9: Hairpinning and shuffling administration

Level	Communication Manager screen	Link to procedure
System	Feature-Related System Parameters	Administering hairpinning and shuffling at the system-level on page 178
Network region	Network Region	Administering hairpinning and shuffling in network regions on page 180
IP Trunks	Signaling Group	Administering H.323 trunks for hairpinning and shuffling on page 183
IP endpoints	Station	Administering IP endpoints for hairpinning and shuffling on page 184

Administering hairpinning and shuffling at the system-level

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

To administer hairpinning and shuffling as a system-level parameter

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

Feature-Related System Parameters screen

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


AUTOMATIC EXCLUSION PARAMETERS

                       Automatic Exclusion by COS? n



                                Recall Rotary Digit: 2

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


    1:         2:         3:         4:         5:

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

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

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

   To disallow shuffled IP calls set this field to **n**. Be sure that you understand the interactions in Hairpinning and shuffling administration interdependencies on page 173 and the notes below.

3. To allow hairpinned audio connections, type **y** (yes) in the **IP Audio Hairpinning** field, noting the interactions in Hairpinning and shuffling administration interdependencies on page 173 and the notes below.

4. Save the changes.

Note:

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

Administering hairpinning and shuffling in network regions

Inter-network region connection management

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

The matrix is used two ways:

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

To administer hairpinning or shuffling within a network region:

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

IP Network Region screen

```
change ip-network-region 1                              Page   1 of   19
                              IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain:
    Name:
                                   Intra-region IP-IP Direct Audio: yes
MEDIA PARAMETERS                   Inter-region IP-IP Direct Audio: yes
   Codec Set: 1                               IP Audio Hairpinning? n
UDP Port Min: 2048
UDP Port Max: 3028                           RTCP Reporting Enabled? n
                                  RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS           Use Default Server Parameters? y
 Call Control PHB Value: 34
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5       AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

2. Administer the **IP-IP Direct Audio** fields:
   - The **Intra-region IP-IP Direct Audio** field permits shuffling if both endpoints are in the same region.

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

The allowable values for both fields are:

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

Note:

If there is no NAT device in use at all, then the native and translated addresses are the same. For more information on NAT, see the *Administrator Guide for Avaya Communication Manager,* 03-300509 and *Avaya Application Solutions: IP Telephony Deployment Guide* (555-245-600).

Note:

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

3. Go to page 3 and administer the common codec sets on the **Inter Network Region Connection Management** screen (Inter Network Region Connection Management screen on page 182). For more detailed information about the fields on this screen, see the Screen Reference chapter of the *Administrator Guide for Avaya Communication Manager,* 03-300509.

Note:

You cannot connect IP endpoints in different network regions or share TN799 C-LAN or TN2032 IP Media Processor resources between/among network regions unless you make a codec entry in this matrix specifying the codec set to be used. For more information, see Administering IP CODEC sets on page 213.

Inter Network Region Connection Management screen

```
change ip-network-region n                                          Page   3 of x

                  Inter Network Region Connection Management

 src dst codec direct    Total       Video                            Dyn
 rgn rgn   set   WAN  WAN-BW-limits Norm Prio Shr Intervening-regions  CAC  IGAR
 3   1     1       y    256:Kbits
 3   2     1       n                          n  1   ___  ___  ___          n
 3   3     1
 3   4     1       n                          y  1   ___  ___  ___          n
 3   5     1       n                          y  6   ___  ___  ___
 3   6     1       y       :NoLimit
 3   7     1       y     10:Calls
 3   8
 3   9     3       y
 3   10
 3   11
 3   12
 3   13
 3   14
 3   15
```

For this example screen, network region 3 communicates with:

- Network regions 1 through 7 using codec set 1
- Network region 9 using codec set 3.

Note:

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

4. Save the changes.

Administering and selecting codecs

When an IP endpoint calls another IP endpoint, Communication Manager asks that the 2nd endpoint choose the same codec that the 1st endpoint offered at call setup. However, if the 2nd endpoint cannot match the 1st's codec, the call is set up with each endpoint's administered (preferred) codec, and the data streams are converted between them, often resulting in degraded audio quality because of the different compressions/decompressions or multiple use of the same codec. For more information, see Administering IP CODEC sets on page 213.

When an endpoint (station or trunk) initially connects to the server, Communication Manager selects the first codec that is common to both the server and the endpoint. The **Inter Network Region Connection Management** screen specifies codec set(s) to use *within* an individual region (intra-region) and a codec set to use *between/among* (inter-region) network regions. Depending upon the network region of the requesting H.323 endpoint or trunk and the network region of the TN2302AP IP Media Processor or TN2602AP IP Media Resource 320 circuit pack:

- If the endpoint and the TN2302AP or TN2602AP are in same region, the administered intra-region codec set is chosen.

- If the endpoint and the TN2302AP or TN2602AP are in different regions, the administered inter-region codec set is chosen.

For example, a region might have its intra-network codec administered as G.711 as the first choice, followed by the other low bit rate codecs. The **Inter Network Region Connection Management** screen for the inter-network region might have G.729 (a low-bit codec that preserves bandwidth) as the only choice. Initially, when a call is set up between these two interconnected regions, the TN2302AP IP Media Processor or TN2602AP IP Media Resource 320 provides the audio stream conversion between G.711 and G.729. When the media stream is shuffled away from a TDM-based connection, the two endpoints can use only the G.729 codec.

Note:

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

Administering H.323 trunks for hairpinning and shuffling

To administer an H.323 trunk for hairpinning or shuffling

1. At the SAT, type **`change signaling group`** ***`number`*** and press **Enter** to display the **Signaling Group** screen (Signaling group screen on page 183).

Signaling group screen

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

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

        Near-end Node Name: mipsn01A          Far-end Node Name: dr98
      Near-end Listen Port: 1800            Far-end Listen Port: 1800
                                         Far-end Network Region:_
              LRQ Required? y             Calls Share IP Signaling Connection? y
              RRQ Required?_
          Media Encryption?_                  Bypass If IP Threshold Exceeded? y
                                                      H.323 Annex H Required?
              DTMF over IP:_                   Direct IP-IP Audio Connections? n
            Link Loss Delay Timer(sec): 90              IP Audio Hairpinning? n
                                             Interworking Message: PROGress
                                       DCP/Analog Bearer Capability: 3.1kHz
```

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

   To disallow shuffled IP calls set this field to **n**. Be sure that you understand the interactions in Hairpinning and shuffling administration interdependencies on page 173 and the notes below.

3. To allow hairpinned audio connections, type **y** (yes) in the **IP Audio Hairpinning** field, noting the interactions in Hairpinning and shuffling administration interdependencies on page 173 and the notes below.

4. Save the changes.

Note:

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

Note:

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

Administering IP endpoints for hairpinning and shuffling

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

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

To administer an IP endpoint for hairpinning or shuffling

1. At the SAT, type **`change station`** ***`extension`*** and press **Enter** to display the **Station** screen (Station screen on page 185)

Station screen

```
change station 57493                                        Page   2 of   4
                                    STATION
FEATURE OPTIONS
           LWC Reception: spe              Auto Select Any Idle Appearance? n
          LWC Activation? y                          Coverage Msg Retrieval? y
  LWC Log External Calls? n                                  Auto Answer: none
            CDR Privacy? n                             Data Restriction? n
   Redirect Notification? y                   Idle Appearance Preference? n
 Per Button Ring Control? n                 Bridged Idle Line Preference? n
   Bridged Call Alerting? n                     Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n          Per Station CPN - Send Calling Number?
      Service Link Mode: as-needed
        Multimedia Mode: basic                    Audible Message Waiting? n
    MWI Served User Type:                       Display Client Redirection? n
            AUDIX Name:                       Select Last Used Appearance? n
                                                Coverage After Forwarding? s
                                                  Multimedia Early Answer? n
                                          Direct IP-IP Audio Connections? y
  Emergency Location Ext: 12345        Always use? n  IP Audio Hairpinning? n
```

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

   To disallow shuffled IP calls set this field to **n**. Be sure that you understand the interactions in Hairpinning and shuffling administration interdependencies on page 173 and the notes below.

3. To allow hairpinned audio connections, type **y** in the **IP Audio Hairpinning** field, noting the interactions in Hairpinning and shuffling administration interdependencies on page 173 and the notes below.

4. Save the changes.

Note:

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

Note:

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

Contradictory IP station administration

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

IP stations used for call center service-observing

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

Administering IP endpoint signal loss

The amount of loss applied between any two endpoints on a call is administrable. However, the Telecommunications Industry Association (TIA) has published standards for the levels that IP endpoints should use. The IP endpoints will always transmit audio at TIA standard levels, and expect to receive audio at TIA standard levels. If an IP audio signal goes to or comes from the TDM bus through a TN2302AP Media Processor or TN2602AP IP Media Resource 320, the circuit pack adjusts the levels to approximately equal the levels of a signal to or from a DCP set. By default, IP endpoints are the same loss group as DCP sets, Group 2.

Adjusting loss to USA DCP levels

The switch instructs the TN2302AP or TN2602AP circuit pack to insert loss into the signal coming from the IP phone, and insert gain in the signal going to the IP phone, to equal the levels of a signal to or from a DCP set.

Note:

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

Note:

The loss that is applied to a hairpinned or shuffled audio connection is constant for all three connection types: station-to-station, station-to-trunk, and trunk-to-trunk

Administering FAX, modem, TTY, and H.323 clear channel calls over IP Trunks

Avaya Communication Manager transports FAX, modem, TTY, and clear channel calls over IP interfaces using relay mode (see What is relay mode on page 187), pass-through mode (see What is pass-through mode on page 188), or both. As a result, Communication Manager supports transport of the following:

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

  Note:

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

  Note:

  Faxes sent to non-Avaya endpoints cannot be encrypted.
- T.38 FAX over the Internet (including endpoints connected to non-Avaya systems)
- Modem tones over a corporate IP intranet
- Clear channel data calls over IP

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

What is relay mode

In relay mode, the firmware on the device (the G700/G350 media gateway, the MM760 VoIP media module, TN2302AP Media Processor, or TN2602AP IP Media Resource 320) detects the tones of the call (FAX, modem, or TTY) and uses the appropriate modulation protocol (for FAX or modem) or Baudot transport representation (TTY) to terminate or originate the call so that it can be carried over the IP network. The modulation and demodulation for FAX and modem calls reduces bandwidth use over the IP network and improves the reliability of transmission. The correct tones are regenerated before final delivery to the endpoint.

Note:

The number of simultaneous calls that a device (gateway, media module, TN2302AP or TN2602AP) can handle is reduced by the modulation and demodulation that the device must perform for relay mode.

What is pass-through mode

In pass-through mode, the firmware on the device (the G700/G350 media gateway, the MM760 VoIP media module, TN2302AP Media Processor, or TN2602AP IP Media Resource 320) detects the tones of the call (FAX, modem, or TTY) and uses G.711 encoding to carry the call over the IP network. pass-through mode provides higher quality transmission when endpoints in the network are all synchronized to the same clock source. The call is un-encoded before final delivery to the endpoint.

Note:

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

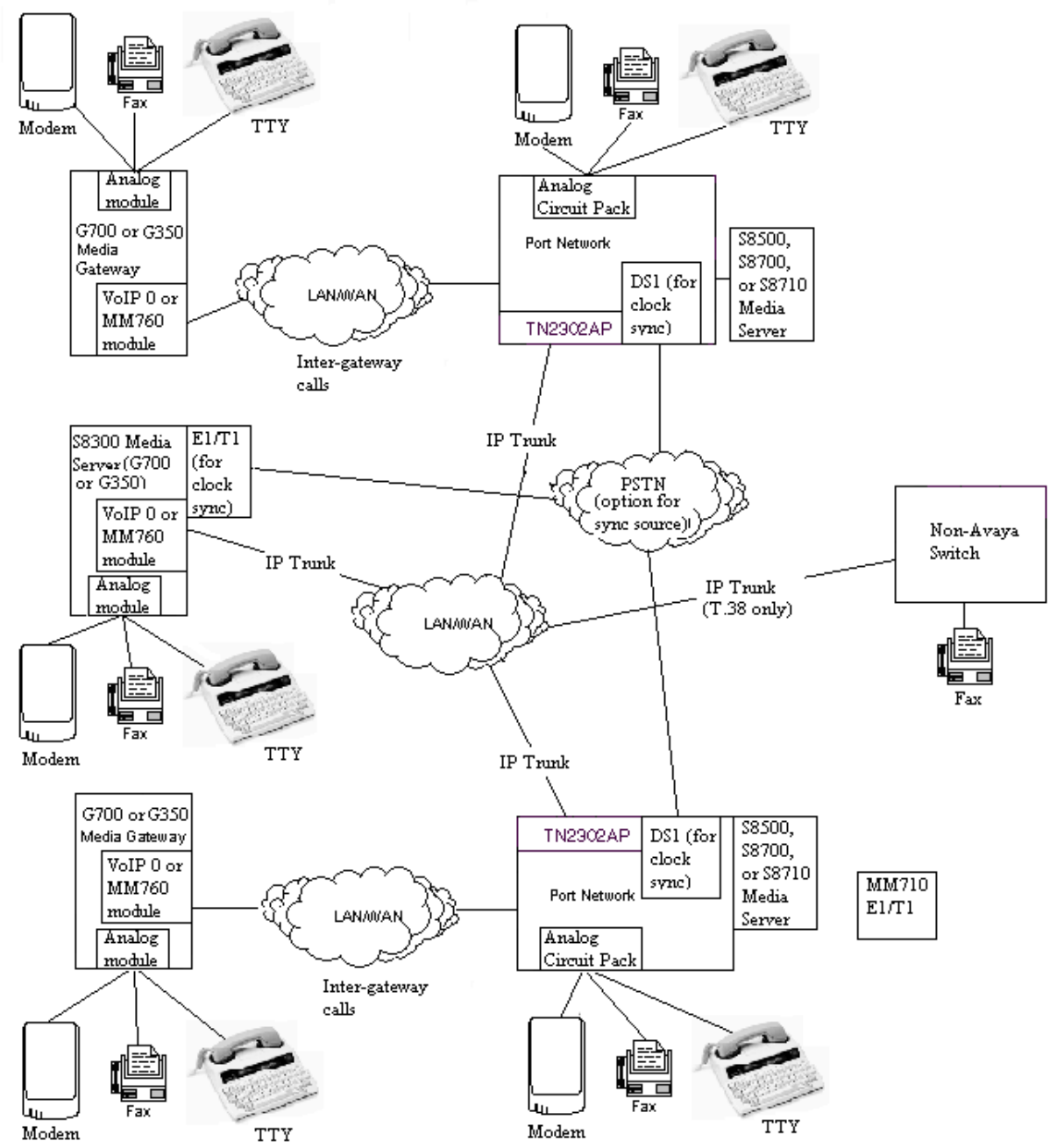
Note:

For pass-through mode on modem and TTY calls over an IP network, the sending and receiving servers should have a common synchronization source. Sychronized clocks can be established by using a source on the public network. See Figure 30: IP network connections over which FAX, modem, and TTY calls are made on page 189.

Note:

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

Figure 30: IP network connections over which FAX, modem, and TTY calls are made



Overview of steps to administer FAX, TTY, modem, and clear channel calls over IP trunks

The information in this section assumes the following:

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

To administer FAX, TTY, modem, and clear channel calls over IP trunks, first consider the following:

- FAX, TTY, modem, and clear channel transmission modes and speeds on page 191
- Considerations for administering FAX, TTY, modem, and clear channel transmission on page 194
- Bandwidth for FAX, modem, TTY, and clear channel calls over IP networks on page 197
- Media encryption for FAX, modem, TTY, and clear channel on page 198

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

1. Create one or more IP Codec sets that enable the appropriate transmission modes for the endpoints on your gateways. See Administering IP CODEC sets on page 213.

   Note:

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

2. Assign each codec set to the appropriate network region. See Administering IP network regions on page 220.
3. Assign the network region to the appropriate device(s):
   - TN2302AP or TN2602AP (see Defining IP interfaces (C-LAN, TN2302AP, or TN2602AP Load Balanced) on page 141)
   - Avaya G350 Media Gateway or Avaya G700 Media Gateway
4. If the TN2302AP or TN2602AP resources are shared among administered network regions, administer inter-network region connections. See Figure 34: IGAR system parameter on page 235.

FAX, TTY, modem, and clear channel transmission modes and speeds

Communication Manager provides the following methods for supporting FAX, TTY, modem, and clear channel transmission over IP (see Table 10: FAX, TTY, modem, and clear channel transmission modes and speeds on page 191).

Table 10: FAX, TTY, modem, and clear channel transmission modes and speeds

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

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Table 10: FAX, TTY, modem, and clear channel transmission modes and speeds (continued)

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

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Table 10: FAX, TTY, modem, and clear channel transmission modes and speeds (continued)

Mode	Maximum Rate	Comments
TTY pass-through	87-110 kbps	In pass-through mode, you can also assign packet redundancy, which means the media gateways send duplicated TTY packets to ensure and improve quality over the network. pass-through mode uses more network bandwidth than relay mode. pass-through TTY uses 87-110 kbps, depending on the packet size, whereas TTY relay uses, at most, the bandwidth of the configured audio codec. Redundancy increases bandwidth usage even more.
Modem Relay	V.32 (9600 bps)	The maximum transmission rate may vary with the version of firmware. The packet size for modem relay is determined by the packet size of the codec selected but is always at least 30ms. Also, each level of packet redundancy, if selected, increases the bandwidth usage linearly (that is, the first level of redundancy doubles the bandwidth usage; the second level of redundancy triples the bandwidth usage, and so on). **Note:** Modem over IP in relay mode is currently available only for use by specific secure analog telephones that meet the Future Narrowband Digital Terminal (FNBDT) standard. See your sales representative for more information. Additionally, modem relay is limited to V.32/V.32bis data rates.
Modem pass-through	V.34 (33.6 kbps) and V.90/V.92 (43.4 kbps)	Transport speed is dependent on the negotiated rate of the modem endpoints. Though the media servers and media gateways support modem signaling at v.34 (33.6 bps) or v.90 and v.92 (43.4 kbps), the modem endpoints may automatically reduce transmission speed to ensure maximum quality of signals. V.90 and V.92 are speeds typically supported by modem endpoints only when directly connected to a service provider Internet service. You can also assign packet redundancy in pass-through mode, which means the media gateways send duplicated modem packets to improve packet delivery and robustness of FAX transport over the network. pass-through mode uses more network bandwidth than relay mode. Redundancy increases bandwidth usage even more. The maximum packet size for modem pass-through is 20 ms.
Clear Channel	64 kbps (unrestricted)	Does not support typically analog data transmission functionality like FAX, modem, TTY, or DTMF signals. It is purely clear channel data. In addition, no support is available for echo cancellation, silence suppression, or conferencing. H.320 video over IP using clear channel is not supported, because of the need for a reliable synchronization source and transport for framing integrity.

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Considerations for administering FAX, TTY, modem, and clear channel transmission

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

- Encryption

  You can encrypt most types of relay and pass-through calls using either the Avaya Encryption Algorithm (AEA) or the Advanced Encryption Standard (AES). See Media encryption for FAX, modem, TTY, and clear channel on page 198.

- Bandwidth usage

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

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

  See Table 11: Bandwidth for FAX, modem, and TTY calls over IP networks on page 197 for the bandwidth usage.

- Calls with non-Avaya systems

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

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

- Differing transmission methods at the sending/receiving endpoints

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

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

- H.320 Video over IP using Clear Channel

  H.320 video is not supported over IP using clear channel, because H.320 video requires a reliable synchronization source and transport for framing integrity of the channels; however, there is no such provision over IP networks. H.320 video might work in some cases for a time, but eventually, the connection would drop because of delay and synchronization problems.

- Hardware requirements

  The relay and pass-through capabilities require the following hardware:

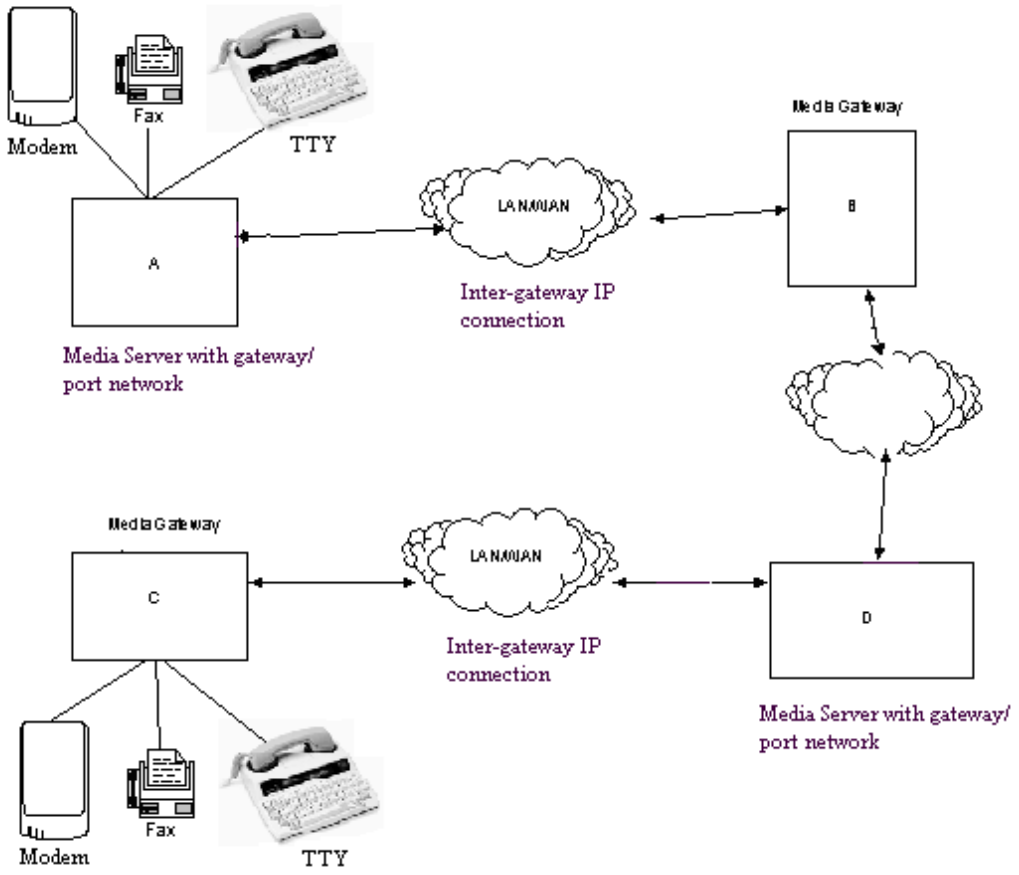
  - For DEFINITY CSI servers, S8500/S8500B Media Servers, or S8700-series Media Servers, certain minimum hardware vintages and firmware versions are required for the TN2302AP or the TN2602AP circuit pack; see the document titled *Avaya Communication Manager Minimum Firmware/Hardware Vintages* at http://www.avaya.com/support.
  - For the G700 or G350 Media Gateway, G700 or G350 firmware version 22.14.0, and VoIP firmware Vintage 40 or greater to support Communication Manager 2.2 is required. An MM760 Media Module with firmware Vintage 40 or greater may be used for additional VoIP capacity. Check the latest firmware on the http://www.avaya.com/support website.
  - For the Avaya S8300/S8300B Media Servers, the Avaya G250 Media Gateway, and the Multi-Tech MultiVoIP Gateway, the firmware should be updated to the latest available on the http://www.avaya.com/support website.
  - For T.38 FAX capability, endpoints on other non-Avaya T.38 compliant communications systems may send FAX calls to or receive FAX calls from endpoints on Avaya systems.

- Multiple hops and multiple conversions

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

  For example, in Figure 31: Shuffling for FAX, modem, and TTY calls over IP on page 196, a hop occurs in either direction for calls between port network A and Media Gateway C because the calls are routed through port network D. In this case, shuffling is required on port network A for calls going to Media Gateway C, and shuffling is required on port network D for calls going from Media Gateway C to port network A.

Figure 31: Shuffling for FAX, modem, and TTY calls over IP



Bandwidth for FAX, modem, TTY, and clear channel calls over IP networks

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

Table 11: Bandwidth for FAX, modem, and TTY calls over IP networks

Packet Size (in msec	Bandwidth (in kbps) (bidirectional)[1]									
	Redundancy = 0						Redundancy = 1		Red. = 2	Red. = 3
	TTY at G.711	TTY at G.729	TTY at G.723[2]	FAX Relay[3]	Modem Relay at 9600 Baud[4]	Clear Channel FAX/Modem pass-through[5] [6]	FAX Relay[3] [4]	Clear Channel FAX/Modem pass-through	FAX Relay[3] [4]	FAX Relay[3] [4]
10	110	54	-	-	-	110	-	221	-	-
20	87	31	-	-	-	87	-	174	-	-
30	79	23	22	25	22.9	-	50	-	75	100
40	76	20	-	-	19.6	-	-	-	-	-
50	73	17	-	-	17.6	-	-	-	-	-
60	72	16	14	-	16.3	-	-	-	-	-

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

Media encryption for FAX, modem, TTY, and clear channel

If media encryption is configured, the algorithm used during the audio channel setup of the call will be maintained for most FAX relay and pass-through modes. The exception is the T.38 standard for FAX over IP, for which encryption is not used.

Note:

Encrypted calls reduce Digital Signal Processing (DSP) capacity by 25% compared to non-encrypted calls.

Encryption is applicable as shown in the following table.

Table 12: Encryption options

Call Type	AEA	AES	SRTP[1]	Transport
Modem Pass-through	Y	Y	Y	RTP (RFC2198)
Modem Relay	Y	N	N	Proprietary
FAX Pass-through	Y	Y	Y	RTP
FAX Relay	Y	(Y)[2]	N	Duplicate Packets
TTY Pass-through	Y	Y	Y	RTP
TTY Relay	Y	Y	Y	RTP
T.38 FAX Standard	(Y)[3]	(Y)[3]	N	T.38 UDPTL Redundancy
Clear Channel	Y	Y	Y	Clear 64 kbps over RTP

1. See SRTP media encryption on page 199 for a description of the SRTP encryption protocol.
2. AES encryption in FAX Relay is available only with Avaya equipment (TN2302) with the correct vintages.
3. The T.38 Fax standard does not support encryption. An enhancement of the T.38 standard enables AES and AEA encryption only with Avaya equipment (TN2302) with the correct vintage.

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

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

The software behaves in the following way for encryption:

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

SRTP media encryption

Secure Real Time Protocol (SRTP) is a media encryption standard that provides encryption of RTP media streams for SIP and 9600-series IP telephones. SRTP is defined in RFC 3711.

The following SRTP features are supported by Communication Manager, release 4.0 and later:

- Encryption of RTP (optional but recommended)
- Authentication of RTCP streams (mandatory)
- Authentication of RTP streams (optional but recommended)
- Protection against replay

The following SRTP features are currently not supported by Communication Manager:

- Encryption of RTCP streams
- Several automatic rekeying schemes
- Various other options within SRTP which are not expected to be used for VoIP, such as key derivation rates or MKIs

Previous releases of Communication Manager supported AEA and AES media encryption for H.323 calls but no media encryption was available for SIP calls. Starting with release 4.0, SRTP provides encryption and authentication of RTP streams for SIP and provides authentication of RTP and RTCP for SIP and H.323 calls using the 9600-series telephones.

SRTP encryption of FAX and modem relay and T.38 is not supported because they are not transmitted in RTP. For this reason, in the case where an SRTP voice call changes to fax relay, fax will not be encrypted.

SRTP is available only if Media Encryption is enabled in the license file and is activated by IP codec set administration in the same manner as for the other encryption algorithms.

Platforms

The SRTP feature is supported on all Linux-based platforms running Communication Manager and on all versions of SES, regardless of platform, starting with the 4.0 release.

The following gateway platforms also support SRTP:

- TN2602AP Media Resource 320
- MM760
- VoIP Media Modules and on-board VoIP engines (G350 and G250).

Administering SRTP

Administering SRTP encryption is the same as administering AES and AEA encryption.

1. Ensure that media encryption is enabled. The Media Encryption? field must be set to ***y*** on the Customer Options form.
2. Administer the Media Encryption type on the ip-codec-set form:

   Media Encryption field — This field appears only if the **Media Encryption over IP** feature is enabled in the license file. Use this field to specify a priority listing of the three possible options for the negotiation of encryption.
3. Administer the ip-network-region form for SIP options:

   Allow SIP URI Conversion? field — Use this field to specify whether a SIP Uniform Resource Identifier (URI) is permitted to change. For example, if "sips://" in the URI is changed to "sip://" then the call would be less secure but this may be necessary to complete the call. If you enter **n** for 'no' URI conversion, then calls made from SIP endpoints that support SRTP to other SIP endpoints that do not support SRTP will fail. Enter **y** to allow conversion of SIP URIs. The default is **y**.

See About Media Encryption on page 246 for more information about administering SRTP.

Chapter 4: Network quality administration

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

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

Note:

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

For more information about QoS in Avaya IP Telephony networks, see *Avaya Application Solutions: IP Telephony Deployment Guide*, 555-245-600.

For more information on implementing QoS, see the White Paper, *Avaya IP Voice Quality Network Requirements (LB1500-02)*, at http://www.avaya.com/master-usa/en-us/resource/assets/whitepapers/lb1500-02.pdf.

About factors causing voice degradation

VoIP applications put severe constraints on the amount of end-to-end transfer delay of the voice signal and routing. If these constraints are not met, users complain of garbled or degraded voice quality, gaps, and pops. Due to human voice perception, VoIP applications can afford to randomly lose a few voice packets and the user can still understand the conversation. However, if voice packets are delayed or systematically lost, the destination experiences a momentary loss of sound, often with some unpleasing artifacts like clicks or pops. Some of the general complaints and their causes are listed in Table 13: User complaints and their causes on page 202.

Table 13: User complaints and their causes

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

Some of the factors causing voice degradation are:

- Packet delay and loss
- Echo
- Transcoding
- Transcoding

Packet delay and loss

The causes of voice degradation include:

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

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

Tip:

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

Echo

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

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

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

Note:

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

Echo cancellers

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

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

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

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

Echo cancellation plans (TN464HP/TN2464CP circuit packs)

The following summarizes the echo cancellation plans that are available exclusively for the TN464HP/TN2464CP circuit packs. For echo cancellation plans that are available for the TN464GP/TN2464BP circuit packs, see Echo cancellation plans (TN464GP/TN2464BP circuit packs) on page 205.

Echo Cancellation Configuration 1 - TN464HP/TN2464CP

This plan is the recommended choice. It has comfort noise generation and residual echo suppression turned on. During "single talk", background noise and residual echo from the distant station may be suppressed and replaced with comfort noise. The comfort noise substitution reduces the perception of background noise pumping, as observed by the talker. In this plan, the EC direction is assumed chosen to cancel the talker's echo. Since this plan turns on comfort noise and echo suppression, it is similar to EC plans 8 and 9 for the TN464GP/TN2464BP circuit packs.

Echo Cancellation Configuration 2 - TN464HP/TN2464CP

This configuration has comfort noise generation turned off and residual echo suppression turned on. This plan may work well in a quiet background environment. In a noisy background environment, background noise pumping/clipping may be heard by the talker. In this case, EC direction is assumed chosen to cancel the talker's echo. This plan my be a good compromise for a small percent of users, who do not care for the comfort noise and prefer the silence during the residual echo suppression periods. Since the plan turns off comfort noise and turns on residual suppression, it is similar to EC configurations 1-6 for the TN464GP/TN2464BP circuit packs.

Echo Cancellation Configuration 3 - TN464HP/TN2464CP

This configuration has comfort noise generation and residual echo suppression turned off. This configuration can be a good choice only if EC plans 1 and 2 do not satisfy the user's preferences. Situations that require configuration 3 should be very rare. (For example, the user does not care for the sound of comfort noise nor the pumping/clipping of background noise.) This configuration allows the user to hear sound from the earpiece as natural as possible. However, the user may hear residual echo during training periods, or all the time if echo is sufficiently high and residual echo is always present. Convergence may be very slow. Since comfort noise and residual suppression are turned off, this configuration is similar to EC configuration 7 for the TN464GP/TN2464BP circuit packs.

Echo cancellation plans (TN464GP/TN2464BP circuit packs)

Communication Manager supports several echo cancellation (EC) plans for the TN464GP/ TN2464BP circuit packs.

Note:

An EC configuration setting can be changed in real time.The change takes effect immediately. That is, it is not necessary to busyout/release the circuit pack – you simply change the setting on the **DS1 Circuit Pack** screen. This can be done without disruption to existing calls - in fact, you immediately hear the effect of the change.

Important:

When there are TN2302AP or TN2602AP circuit pack(s) and TN464GP/ TN2464BP circuit pack(s) being used for a call, the echo canceller on the TN2302AP or TN2602AP is turned off and the echo canceller on the TN464GP/ TN2454BP is used instead, because it has the greater echo canceller.

The following summarizes the echo cancellation plans that are available for the TN464GP/ TN2464BP circuit packs. For echo cancellation plans that are available exclusively for the TN464HP/TN2464CP circuit packs, see Echo cancellation plans (TN464HP/TN2464CP circuit packs) on page 204.

Echo Cancellation Configuration 1 – Highly Aggressive Echo Control

This configuration can control very strong echo from a distant party. It (as well as Echo Cancellation Configuration 4) provides the most rapid convergence in detecting and correcting echo at the beginning of a call. The initial echo fades faster than the other settings (generally in a small fraction of a second), regardless of the loudness of the talker's voice. EC Configurations 1 and 4 are the same except for loss. EC Configuration 1 has 6dB of loss and EC 4 has 0dB of loss. This makes EC Configuration 1 a good choice for consistently high network signal levels. EC Configuration 1 can cause low-volume complaints and/or complaints of clipped speech utterances, particularly when both parties speak simultaneously (doubletalk). Because EC Configuration 1 relies strongly on echo suppression to help control echo, "pumping" of the distant party's background noise may occur and lead to complaints. Prior to Communication Manager Release 2.0, EC Configuration 1 was the default configuration.

The 6dB of loss in EC Configuration 1 is in one direction only and depends on the setting of the **EC Direction** field on the **DS1 Board** screen. If the direction is set to **inward**, then the 6dB of loss is inserted in the path out from the board towards the T1/E1 circuit. Conversely, if the setting is **outward**, then the 6dB of loss is inserted into the path from the T1/E1 circuit towards the TDM bus.

Echo Cancellation Configuration 2 – Aggressive, Stable Echo Control

This configuration is nearly identical to EC Configuration 1, except that it does not inject an additional 6dB of signal loss, *and* convergence of the echo canceller is slower, but more stable than that provided by EC Configuration 1. If EC Configuration 1 is found to diverge during doubletalk conditions – noticeable by the sudden onset of audible echo, EC Configuration 2 should be used in place of EC Configuration 1. Because the echo canceller converges somewhat slower, some initial echo may be noticeable at the start of a call, while the system is "training". EC Configuration 2 can cause complaints of clipped speech utterances, particularly during doubletalk. Because EC Configuration 2 relies strongly on echo suppression to help control echo, "pumping" of the distant party's background noise may occur and lead to complaints.

Echo Cancellation Configuration 3 – Aggressive, Very Stable Echo Control

This configuration is nearly identical to EC Configuration 2, but is even more stable. Because the echo canceller converges somewhat slower, some initial echo may be noticeable at the start of a call. EC Configuration 3 can cause complaints of clipped speech utterances, particularly during doubletalk. Because EC Configuration 3 relies strongly on echo suppression to help control echo, "pumping" of the distant party's background noise may occur and lead to complaints.

Echo Cancellation Configuration 4 – Highly Aggressive Echo Control

Echo Cancellation Configuration 4 is identical to EC Configuration 1, but does not provide the 6dB loss option as described for EC Configuration 1. All other comments from EC Configuration 1 apply to EC Configuration 4. EC Configuration 4 can cause complaints of clipped speech utterances, particularly during doubletalk. Because EC Configuration 4 strongly relies on echo suppression to help control echo, “pumping” of the distant party’s background noise may occur, and lead to complaints.

Echo Cancellation Configuration 5 – Very Moderate, Very Stable Echo Control

Echo Cancellation Configuration 5 departs significantly from EC Configurations 1 –4. The echo canceller is slower to converge and is very stable once it converges. Some initial echo may be heard at the beginning of a call. EC Configuration 5 will not, in general, lead to complaints of clipped speech or pumping of the distant party’s background noise.

Echo Cancellation Configuration 6 – Highly Aggressive Echo Control

Echo Cancellation Configuration 6 is identical to EC Configuration 4, but reliance on the echo suppressor to control echo is about one-half that of EC Configuration 4. As a result, EC Configuration 6 will not clip speech as much as EC Configuration 4, but may cause somewhat more audible echo, particularly at the start of a call. Some pumping of the distant party’s background noise may be perceptible.

Echo Cancellation Configuration 7 – Extremely Moderate & Stable Echo Control

Echo Cancellation Configuration 7 provides very stable and transparent control of weak to low-level echoes. For connections having audible echo at the start of a call, the residual echo may linger for several seconds as the echo canceller converges.

Echo Cancellation Configuration 8 –Aggressive, Very Transparent Echo Control 1

Echo Cancellation Configuration 8 provides aggressive control of echo at the start of a call and more moderate control during the call. Unlike all prior settings, EC Configuration 8 uses “comfort noise” injection to match the actual noise level of the distant party’s speech signal. The effect is one of echo canceller “transparency,” in which complaints of clipped speech or noise pumping should be few to none. To many people, EC Configuration 8 and EC Configuration 9 will be indistinguishable.

Echo Cancellation Configuration 9 – Aggressive, Transparent Echo Control 2

Echo Cancellation Configuration 9 is nearly identical to EC Configuration 8, but provides somewhat more residual echo control at a slight expense of transparency. To many people, EC Configuration 8 and EC Configuration 9 will be indistinguishable.

Transcoding

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

Bandwidth

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

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

About Quality of Service (QoS) and voice quality administration

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

For example, when delay is minimized:

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

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

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

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

These parameters are administered on the **IP Network Region** screen (see Administering IP network regions on page 220).

Layer 3 QoS

DiffServ

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

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

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

RSVP

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

Layer 2 QoS: 802.1p/Q

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

CAUTION:

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

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

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

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

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

Using VLANs

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

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

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

- Circuit pack-level administration on the **IP-Interfaces** screen (see Defining IP interfaces (C-LAN, TN2302AP, or TN2602AP Load Balanced) on page 141)
- Endpoint-level administration on the **IP Address Mapping** screen

To administer endpoints for IP address mapping

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

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

                      IP ADDRESS MAPPING

                                                                  Emergency
                                        Subnet          802.1Q    Location
FROM IP Address   (TO IP Address or Mask)   Region      VLAN      Extension
  1.__2.__3.__0    1.__2.__3.255    24      __1         ___3      _________
  1.__2.__4.__4    1.__2.__4.__4    32      __2         ___0      _________
  1.__2.__4.__5    1.__2.__4.__5    __      __3         ___0      _________
  1.__2.__4.__6    1.__2.__4.__9    __      __4         ___4      _________
___.___.___.___  ___.___.___.___    __      ___         ____      _________
___.___.___.___  ___.___.___.___    __      ___         ____      _________
___.___.___.___  ___.___.___.___    __      ___         ____      _________
___.___.___.___  ___.___.___.___    __      ___         ____      _________
___.___.___.___  ___.___.___.___    __      ___         ____      _________
___.___.___.___  ___.___.___.___    __      ___         ____      _________
___.___.___.___  ___.___.___.___    __      ___         ____      _________
___.___.___.___  ___.___.___.___    __      ___         ____      _________
___.___.___.___  ___.___.___.___    __      ___         ____      _________
```

2. Complete the following fields:

Table 14: IP Address Mapping screen fields

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

1 of 2

Table 14: IP Address Mapping screen fields (continued)

Field	Conditions/Comments
or Subnet Mask	Specifies the mask to be used to obtain the subnet work identifier from the IP address. If this field is non-blank on submission, then: • Mask applied to **From IP Address** field, placing zeros in the non-masked rightmost bits. This becomes the stored "From" address. • Mask applied to **To IP Address** field, placing 1's in the non-masked rightmost bits. This becomes the stored "To" address. If this field and the **To IP Address** field are blank when submitted, the address in the **From IP Address** field is copied into the **To IP Address** field. Valid entries: **0-32**, or blank.
Region	Identifies the network region for the IP address range. Valid entries: **1-250** (Enter the network region number for this interface.)
VLAN	Sends VLAN instructions to IP endpoints such as IP telephones/IP Softphones. This field does not send instructions to the PROCR, C-LAN, or Media Processor boards. Valid entries: **0-4095** (specifies the virtual LAN value); **n** (disabled).
Emergency Location Extension	Enter a value of 1-7 digits in length for the emergency location extension. Default is blank. (A blank entry typically would be used for an IP softphone dialing in through PPP from somewhere outside your network.) If the entry on this screen differs from the value entered in the **Emergency Location Extension** field on the **Station** screen, then it is the extension entered on this screen that will be sent to the Public Safety Answering Point (PSAP).

2 of 2

3. Submit the screen.

Administering IP CODEC sets

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

Note:

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

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

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

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

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

CAUTION:

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

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

To administer an IP Codec set

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

IP Codec Set screen, Page 1

```
change ip-codec-set 1                                    Page 1 of 2
                         IP CODEC SET
Codec Set: 1
    Audio    Silence       Frames   Packet
    Codec    Suppression   per Pkt  Size (ms)
1.  G.711mu       n          2        20
2.  G.729         n          2        20
3.  G.711mu       y          2        20
4.
5.
6.
7.
Media Encryption:
1: aes
2: aea
3: 1-srtp-aescm128-hmac80
```

2. Complete the fields in Table 15:

Note:

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

- G.711 A-law — 64Kbps
- G.711 mu-law — 64Kbps (used in U.S. and Japan)
- G.729 — 8 kbps
- G.729A/B/AB — 8 kbps audio

Table 15: IP Codec Set screen fields, page 1

Field	Conditions/Comments
Audio Codec	Specifies an audio CODEC. Valid values are: • **G.711A** (a-law) • **G.711MU** (mu-law) • **G.722- 64k** • **G.722.1- 24k** • **G.722.1- 32k** • **G.723- 5.3k** • **G.723- 6.3k** • **G.726A- 32k** • **G.729** • **G.729A** • **G.729B** • **G.729AB** • **SIREN14- 24k** • **SIREN14- 32k** • **SIREN14- 48k** • **SIREN14- S48k** • **SIREN14- S56k** • **SIREN14- S64k** • **SIREN14- S96k**
Silence Suppression	Enter **n** (recommended). Enter **y** if you require silence suppression on the audio stream. This may affect audio quality.
Frames per Pkt	Specifies frames per packet. Enter a value between **1-6**. Default values are: • **2** for G.711 Codec (frame size 10ms) • **2** for G729 Codec (frame size 10ms)
Packet Size (ms)	Automatically appears.

1 of 2

Table 15: IP Codec Set screen fields, page 1 (continued)

Field	Conditions/Comments
Media Encryption	This field appears only if the **Media Encryption over IP** feature is enabled. It specifies one of three possible options for the negotiation of encryption. The selected option for an IP codec set applies to all codecs defined in that set. Valid entries are: • **aes** — Advanced Encryption Standard (AES), a standard cryptographic algorithm for use by U.S. government organizations to protect sensitive (unclassified) information. Use this option to encrypt these links: - Server-to-gateway (H.248) - Gateway-to-endpoint (H.323) • **aea** — Avaya Encryption Algorithm. Use this option as an alternative to AES encryption when: - All endpoints within a network region using this codec set must be encrypted. - All endpoints communicating between two network regions and administered to use this codec set must be encrypted. • **1-srtp-aescm128-hmac80** — Encrypted/Authenticated RTP with 80-bit authentication tag • **2-srtp-aescm128-hmac32** — Encrypted/Authenticated RTP with 32-bit authentication tag • **3-srtp-aescm128-hmac80-unauth** — Encrypted RTP but not authenticated • **4-srtp-aescm128-hmac32-unauth** — Encrypted RTP but not authenticated • **5-srtp-aescm128-hmac80-unenc** — Authenticated RTP with 80-bit authentication tag but not encrypted • **6-srtp-aescm128-hmac32-unenc** — Authenticated RTP with 32-bit authentication tag but not encrypted • **7-srtp-aescm128-hmac80-unenc-unauth** — Unencrypted/ Unauthenticated RTP • **8-srtp-aescm128-hmac32-unenc-unauth** — Unencrypted/ Unauthenticated RTP • **none** — Media stream is unencrypted. This is the default setting.

2 of 2

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

   Page 2 appears.

IP-Codec-Set, page 2

```
change ip-codec-set n                                    Page   2 of   x

                            IP Codec Set

                                Allow Direct-IP Multimedia? y
       Maximum Bandwidth Per Call for Direct-IP Multimedia: 256:Kbits

                    Mode         Redundancy

            FAX     relay          0

          Modem     off            0

        TDD/TTY     us             0

  Clear-channel     n              0
```

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

Table 16: IP Codec Set screen fields, page 2

Field	Conditions/Comments
All Direct-IP Multimedia?	Enter **y** to allow direct multimedia via the following codecs: • H.261 • H.263 • H.264 (video) • H.224 H.224.1 (data, far-end camera control).
Maximum Bandwidth Per Call for Direct-IP Multimedia	This field displays only when **Allow Direct-IP Multimedia** is **y**. Enter the unit of measure, **kbits** or **mbits**, corresponding to the numerical value entered for the bandwidth limitation. Default is **kbits**

1 of 3

Table 16: IP Codec Set screen fields, page 2 (continued)

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

Table 16: IP Codec Set screen fields, page 2 (continued)

Field	Conditions/Comments
TDD/TTY Mode	Specifies the mode for TDD/TTY calls. Valid values are: • **off** Turn off special TTY handling when using this codec set. In this case, the TTY transmission is treated like an ordinary voice call. With a codec set that uses G.711, this setting is required to send TTY calls to non-Avaya systems. However, there may be errors in character transmissions. • **US** For users in regions using this codec, use U.S. Baudot 45.45 mode for TTY transmissions over IP network facilities. This is the default for new installations and upgrades to Communication Manager R2.1. • **UK** For users in regions using this codec, use U.K. Baudot 50 mode for TTY transmissions over IP network facilities. • **pass-through** For users in regions using this codec, use pass-through mode for TTY transmissions over IP network facilities.
Clear Channel	• **"y"**es allows 64 kbps clear channel data calls for this codec set. • **"n"**o disallows 64 kbps clear channel data calls for this codec set.
Redundancy	For each type of call (TTY, fax, modem, or clear channel) that does *not* use pass-through mode, enter the number of duplicated packets, from **0** to **3**, that the system sends with each primary packet in the call. **0** means that you do not want to send duplicated packets. For any call types for which you selected pass-through or clear channel modes, you can enter **0** or **1** only. That is, for pass-through and clear channel modes, the maximum number of duplicated packets that the system can send with each primary packet is one.

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5. Submit the screen.

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

Codec Sets screen

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

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

7. Review your CODEC sets.

Administering IP network regions

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

- Audio Parameters
  - Codec Set
  - UDP port Range
  - Enabling Direct IP-IP connections
  - Enabling Hairpinning
- Quality of Service Parameters:
  - Diffserv settings
    - Call Control per-hop behavior (PHB)
    - VoIP Media PHB
  - 802.1p/Q settings
    - Call Control 802.1p priority
    - VoIP Media 802.1p priority
    - VLAN ID
  - Better than Best Effort (BBE) PHB
  - RTCP settings
  - RSVP settings
  - Location

- WAN bandwidth limitations
  - Call Admission control - Bandwidth Limitation (CAC-BL)
  - Inter-Gateway Alternate Routing (IGAR)

The following sections tell you about:

- Defining an IP network region
- Setting up Inter-Gateway Alternate Routing (IGAR)
- Setting up Dial Plan Transparency
- Network Region Wizard (NRW)
- Manually interconnecting the network regions
- Administering inter-network region connections
- Pair-wise administration of IGAR between network regions
- Reviewing the network region administration

Note:

For more information on using network regions, with examples, see the application note *Network Regions for Avaya MultiVantage™ Solutions - A Tutorial*, which is available at: http://www.avaya.com/gcm/master-usa/en-us/resource/assets/applicationnotes/advantages_of_implem.pdf (requires Adobe Reader). For more information on configuring network regions in Avaya Communication Manager, see the application note *Avaya Communication Manager Network Region Configuration Guide*, which is available at: http://www.avaya.com/master-usa/en-us/resource/assets/applicationnotes/netw-region-tutorial.pdf (requires Adobe Reader).

Defining an IP network region

CAUTION:

Never define a network region to span a WAN link.

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

To define an IP network region

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

IP Network Region screen

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

                               IP NETWORK REGION
  Region: 1
Location:                  Authoritative Domain:
    Name:
                                Intra-region IP-IP Direct Audio: no
AUDIO PARAMETERS                Inter-region IP-IP Direct Audio: no
   Codec Set: 1                           IP Audio Hairpinning? n
UDP Port Min: 2048
UDP Port Max: 3049                      RTCP Reporting Enabled? y
                              RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS       Use Default Server Parameters? n
 Call Control PHB Value: 46                 Server IP Address:   .   .   .
        Audio PHB Value: 46                       Server Port: 5005
802.1P/Q PARAMETERS                    RTCP Report Period(secs): 5
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 7
                                      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                    RSVP Enabled? y
   H.323 Link Bounce Recovery? y               RSVP Refresh Rate(secs) 15
  Idle Traffic Interval (sec): 20     Retry upon RSVP Failure Enabled? y
    Keep-Alive Interval (sec): 6                      RSVP Profile:
            Keep-Alive Count: 5      RSVP unreserved (BBE) PHB Value: 40
```

2. Complete the fields using the information in Table 17: IP Network Region field descriptions on page 222.

Table 17: IP Network Region field descriptions

Field	Descriptions/Comments
Region	Network Region number, **1–250**.
Location	Blank or **1–250**. Enter the number for the location for the IP network region. The IP endpoint uses this as its location number. This applies to IP telephones and IP Softphones. **1-44** (DEFINITY CSI) **1-250** (S8300, S8500, S8700, S8710, S8720 Media Servers) **blank** The location is obtained from the cabinet containing the C-LAN that the endpoint registered through, or the media gateway containing the Internal Call Controller or Local Survivable Processor on an Avaya S8300 Media Server through which the endpoint registered. This applies to IP telephones and IP Softphones. Traditional cabinets, Remote Offices, and the Avaya S8300 Media Server all have their locations administered on their corresponding screens.

Table 17: IP Network Region field descriptions (continued)

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

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Table 17: IP Network Region field descriptions (continued)

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

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Table 17: IP Network Region field descriptions (continued)

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

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Table 17: IP Network Region field descriptions (continued)

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

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CAUTION:

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

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

Call Admission Control

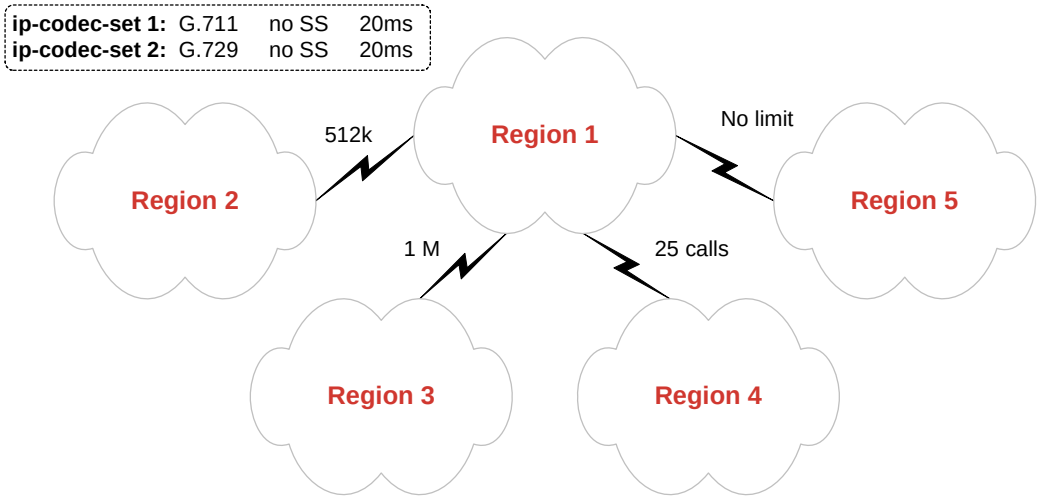
Call Admission Control (CAC) is a feature that allows a limit to be set on the bandwidth consumption or number of calls between network regions.

Note:

If SRTP media encryption is used for SIP and H.323 calls, CAC must be adjusted for the additional overhead imposed by the authentication process. SRTP authentication can add 4 (HMAC32) or 10 (HMAC80) bytes to each packet.

The primary use of this feature is to prevent WAN links from being overloaded with too many calls. This is done by setting either a bandwidth limit or a number-of-calls limit between network regions, as follows:

- Bandwidth consumption is calculated using the methodology explained in the *Avaya Application Solutions: IP Telephony Deployment Guide* (555-245-600).
- The L2 overhead is assumed to be 7 bytes, which is the most common L2 overhead size for WAN protocols.
- The calculated bandwidth consumption is rounded up to the nearest whole number.
- The calculated bandwidth consumption takes into account the actual IP CODEC being used for each individual call. It does not assume that all calls use the same CODEC.
- If the administrator chooses not to have the media server calculate the bandwidth consumption, he/she may enter in a manual limit for the number of calls. However, this manually entered limit is adhered to regardless of the codec being used. Therefore, the administrator must be certain that either all calls use the same CODEC, or that the manual limit takes into account the highest possible bandwidth consumption for the specified inter-region CODEC set.
- If a call between two network regions traverses an intervening network region (for example, a call from 1 to 3 actually goes 1 to 2 to 3), then the call server keeps track of the bandwidth consumed across both inter-region connections, that is, both 1 to 2 and 2 to 3.



The figure above shows a simple hub-spoke network region topology. The WAN link between network regions 1 and 2 has 512kbps reserved for VoIP. The WAN link between network regions 1 and 3 has 1Mbps reserved for VoIP. The link between network regions 1 and 4 is one where the 7-byte L2 overhead assumption would not hold, such as an MPLS or VPN link. In this case, the administration is such that all inter-region calls terminating in region 4 use the G.729 codec (with no SS at 20ms).

Therefore, it is feasible to set a limit on the number of inter-region calls to region 4, knowing exactly how much bandwidth that CODEC consumes (with the MPLS or VPN overhead added). Finally, the link between network regions 1 and 5 requires no limit, either because there are very few endpoints in region 5 or because there is practically unlimited bandwidth to region 5.

The corresponding **IP Network Region** screens for each network region are shown below.

Configure inter-region connectivity for network region 1.

```
change ip-network-region 1                                    Page   3 of  19

                  Inter Network Region Connection Management

 src dst  codec  direct                                    Dynamic CAC
 rgn rgn   set    WAN    WAN-BW-limits  Intervening-regions   Gateway    IGAR
 1   1     1
 1   2     2       y        512:Kbits
 1   3     2       y          1:Mbits
 1   4     2       y         25:Calls
 1   5     2       y            :NoLimit
```

- Connectivity from network region 1 to all the other regions is configured per the diagram above.
- All the inter-region connections use the WAN codec set.

Configure inter-region connectivity for network region 2.

```
change ip-network-region 2                                    Page   3 of  19

                  Inter Network Region Connection Management

 src dst  codec  direct                                    Dynamic CAC
 rgn rgn   set    WAN    WAN-BW-limits  Intervening-regions   Gateway    IGAR
 2   1     2       y        512:Kbits
 2   2     1
 2   3     2       n                         1
 2   4     2       n                         1
 2   5     2       n                         1
```

- Network region 2 connects to regions 3, 4, and 5 via intervening region 1.
- Communication Manager keeps track of the bandwidth or call limits between all adjacent regions.

Configure inter-region connectivity for network region 3.

```
change ip-network-region 3                                    Page   3 of  19

                  Inter Network Region Connection Management

 src dst  codec  direct                                    Dynamic CAC
 rgn rgn   set    WAN    WAN-BW-limits  Intervening-regions   Gateway    IGAR
 3   1     2       y          1:Mbits
 3   2     2       n                         1
 3   3     1
 3   4     2       n                         1
 3   5     2       n                         1
```

The corresponding **IP Network Region** screens for each network region are shown below.

Configure inter-region connectivity for network region 4.

```
change ip-network-region 4                                    Page   3 of  19

                    Inter Network Region Connection Management

 src dst  codec  direct                                    Dynamic CAC
 rgn rgn   set    WAN    WAN-BW-limits  Intervening-regions   Gateway    IGAR
 4   1     2       y         25:Calls
 4   2     2       n                          1
 4   3     2       n                          1
 4   4     1
 4   5     2       n                          1
```

Configure inter-region connectivity for network region 5.

```
change ip-network-region 5                                    Page   3 of  19

                    Inter Network Region Connection Management

 src dst  codec  direct                                    Dynamic CAC
 rgn rgn   set    WAN    WAN-BW-limits  Intervening-regions   Gateway    IGAR
 5   1     2       y           :NoLimit
 5   2     2       n                          1
 5   3     2       n                          1
 5   4     2       n                          1
 5   5     1
```

Setting up Inter-Gateway Alternate Routing (IGAR)

Whenever Communication Manager needs an inter-gateway connection and sufficient IP bandwidth is not available, it attempts to substitute a trunk connection for the IP connection. This happens in any of a large variety of scenarios, including the following examples:

- A party in one Network Region (NR) calls a party in another NR, or
- A station in one NR bridges onto a call appearance of a station in another NR, or
- An incoming trunk in one NR routes to a hunt group with agents in another NR, or
- An announcement or music source from one NR must be played to a party in another NR.

Communication Manager software automatically attempts to use a trunk for inter-region voice bearer connection when *all* of the following five conditions are met:

- An inter-gateway connection is needed.
- IGAR has been “triggered” by one (or more) of the following conditions:
  - The administered bandwidth limit between two NRs has been exhausted, or
  - The VoIP resources between two PN/MGs have been exhausted, or
  - IGAR has been “forced” between two NRs, or
  - The codec set is set to pstn.
- IGAR is enabled for the NRs associated with each end of the call.
- The System Parameter **Enable Inter-Gateway Alternate Routing** is set to ‘y’. See Figure 34.
- The number of trunks used by IGAR in each of the two NRs has not reached the limit administered for that NR.

A Trunk IGC is established using ARS to route a trunk call from one NR to the *IGAR LDN Extension* administered for other NR. Because the Trunk IGC is independent of the actual call being placed, Communication Manager can originate the IGC in either direction — that is, from the calling party’s NR to the NR of the called party, or vice versa. However, because some customers wish to use Facility Restriction Levels or Toll Restriction to determine who gets access to IGAR resources during a WAN outage, the calling user is considered the originator of the Trunk IGC for the purposes of authorization (for example, FRL checking) and routing (for example, determining which Location-specific ARS and Toll tables to use). However, if the outgoing trunk group is administered to send the Calling Number, the *IGAR Extension* in the originating NR is used to create this number using the appropriate administration (performed on the public/unknown or private numbering screen).

The following are examples of certain failure conditions and how Communication Manager handles them:

- On a direct call, the call proceeds to the first coverage point of the unreachable called endpoint, or if no coverage path is assigned, busy tone is played to the calling party.
- If the unreachable endpoint is being accessed through a coverage path, it is skipped.
- If the unreachable endpoint is the next available agent in a hunt group, that agent is considered unavailable, and the system tries to terminate to another agent using the administered group type (Circular, Percent Allocation Distribution, etc.).

Setting up Dial Plan Transparency

Dial Plan Transparency (DPT) preserves the dial plan when a media gateway registers with an LSP or when a port network registers with an ESS due to the loss of contact with the primary controller. In this scenario, DPT establishes a trunk call and reroutes the call over the PSTN to connect endpoints that can no longer connect over the corporate IP network.

DPT does not need to be activated in the license file. DPT is available as a standard feature for Communication Manager Release 4.0 and later.

DPT is similar to IGAR in that they both provide alternate routing of calls when normal connections are not available. A major difference between DPT and IGAR is that DPT routes calls between endpoints controlled by two independent servers while IGAR routes calls between endpoints controlled by a single server. The DPT and IGAR features are independent of each other but can be activated at the same time.

Limitations of DPT include the following:

- DPT only handles IP network connectivity failures *between* network regions.
- Because DPT calls are trunk calls, many station features are not supported.
- For Release 4.0, DPT applies only to endpoints that are dialed directly. Redirected calls or calls to groups cannot be routed by DPT.
- DPT cannot reroute calls involving a SIP endpoint that has lost registration with its Home SES.
- Failover strategies for gateways and port networks, and alternate gatekeeper lists for IP stations, must be kept consistent for DPT to work.

Use the following procedure to administer DPT

1. Enable DPT on the Feature-Related System Parameters screen
   a. set **Enable Dial Plan Transparency in Survivable Mode** to **y**.
   b. Set **COR to Use for DPT** to either **station** or **unrestricted**.

      If set to **station**, the Facility Restriction Level (FRL) of the calling station determines whether that station is permitted to make a trunk call and if so, which trunks it is eligible to access. If set to **unrestricted**, the first available trunk preference pointed to by ARS routing is used.

Figure 32: Enabling DPT on the System Features screen

```
change system -parameters features                                  Page 5 of x
                        FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
  Endpoint: 24099          Lines Per Page: 40

SYSTEM-WIDE PARAMETERS
                                      Switch Name: Mercury
             Emergency Extension Forwarding (min): 4
           Enable Inter-Gateway alternate Routing? n
  Enable Dial Plan Tranparency in Survivable Mode? y
                                COR to Use for DPT: station

MALICIOUS CALL TRACE PARAMETERS
                Apply MCT Warning Tone? n   MCT Voice Recorder Trunk Group:
       Delay SEnding RELease (seconds)? 0
SEND ALL CALLS OPTIONS
      Send All Calls Applies to: extension  Auto Inspect on Send All Calls? n

UNIVERSAL CALL ID
       Create Universal Call ID (UCID)? y    UCID Network Node ID: 10
```

2. Enable DPT for the appropriate Network Regions. On page 2 of the IP Network Region screen, set the **Dial Plan Transparency in Survivable Mode** field to **y**.

Figure 33: Enabling DPT on the Network Region screen

```
change ip-network-region 1                                    Page 2 of 19
                                IP NETWORK REGION

INTER-GATEWAY ALTERNATE ROUTING / DIAL PLAN TRANSPARENCY
 Incoming LDN Extension: 852-3999
 Conversion To Full Public Number - Delete: 0   Insert: +1732________
  Maximum Number of Trunks To Use for IGAR: 23
 Dial Plan Transparency in Survivable Mode? y

 BACKUP SERVERS(IN PRIORITY ORDER)          H.323 SECURITY PROFILES
 1   __________________                     1   challenge
 2   __________________                     2
 3   __________________                     3
 4   __________________                     4
 5   __________________
 6   __________________                     Allow SIP URI Conversion? y

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
   Near End Establishes TCP Signaling Socket? n
                        Near End TCP Port Min: 61440
                        Near End TCP Port Max: 61444
```

3. If not already completed for IGAR, allocate on incoming DID / LDN extension for incoming DPT calls. This extension can be shared by IGAR and DPT.
4. As for IGAR, ensure that a sufficient number of trunks are available. You do not need to set the maximum number of trunks for DPT.
5. Use existing routing techniques to ensure that an outgoing DPT call from a given Network Region has access to an outgoing trunk. The outgoing trunk need not be in the same Network Region as the calling endpoint, as long as the endpoint and trunk Network Regions are interconnected.

Network Region Wizard (NRW)

The Avaya Network Region Wizard (NRW) is a browser-based wizard that is available on Avaya Media Servers running Communication Manager 2.1 or higher software. The NWR supports IGAR along with prior support for CAC and codec set selection for inter-connected region pairs. For any system that has several network regions, the use of the wizard can save time for the software specialist or business partner provisioning the system, as well as help to configure the system for optimum IP performance.

The NRW guides you through the steps required to define network regions and set all necessary parameters through a simplified, task-oriented interface. The purpose of the NRW is to simplify and expedite the provisioning of multiple IP network regions, including Call Admission Control via Bandwidth Limits (CAC-BL) for large distributed single-server systems that have several network regions. The NRW is especially valuable for provisioning systems with dozens or hundreds of network regions, for which administration using the System Access Terminal (SAT) scales poorly.

NRW provisioning tasks include:

- Specification and assignment of codec sets to high-bandwidth (intra-region) LANs and lower-bandwidth (inter-region) WANs
- Configuration of IP network regions, including all intra-region settings, as well as inter-region administration of CAC-BL for inter-region links
- Ongoing network region administration by the customer as well as by Avaya technicians and Business Partners to accommodate changes in the customer network following cutover
- Assignment of VoIP resources (C-LANs, TN2302/TN2602 circuit packs, Media Gateways), and endpoints to IP network regions.

The NRW simplifies and expedites network region provisioning in several ways:

- NRW uses algorithms and heuristics based on graph theory to greatly reduce the repetitive manual entry required by the SAT to configure codecs, and CAC-BL for inter-region links. With the SAT, the number of inter-region links that need to be configured by the user does not scale well; with the NRW, the number of region pairs that require manual administration will increase *linearly* with the number of regions.

- NRW provides templates of widely applicable default values for codec sets and intra-region parameter settings. Users have the ability to customize these templates with their own default values.
- NRW runs on any Internet browser supported by the Avaya Integrated Management (IM) product line, and takes advantage of browser capabilities to offer user-friendly prompting and context-sensitive online help.

The NRW has its own Job Aid and worksheet (one of Avaya's wizard tools that are available from http://support.avaya.com/avayaiw), and is a standard IM support tool delivered with every Linux-based Communication Manager system.

Manually interconnecting the network regions

Use the **Enable Inter-Gateway Alternate Routing**? field on the Feature-Related System Parameters screen to enable IGAR on a system-wide basis. Using this parameter, IGAR can be quickly disabled without changing/removing other feature administration associated with IGAR. This parameter is included under the **System-Wide Parameters**, as shown in Figure 34.

Figure 34: IGAR system parameter

```
change system-parameters features                              Page 5 of 14
                        FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
  Endpoint: SYS_PRNT       Lines Per Page: 60

SYSTEM-WIDE PARAMETERS
                                      Switch Name: Skipper
            Emergency Extension Forwarding (min): 10
          Enable Inter-Gateway Alternate Routing? n
Enable Dial Plan Transparency in Survivable Mode? y
                              COR to Use for DPT: station

MALICIOUS CALL TRACE PARAMETERS
               Apply MCT Warning Tone? y   MCT Voice Recorder Trunk Group: 256
      Delay Sending RELease (seconds)? 0
SEND ALL CALLS OPTIONS
     Send All Calls Applies to: station    Auto Inspect on Send All Calls? n

UNIVERSAL CALL ID
     Create Universal Call ID (UCID)? y    UCID Network Node ID: 10040
```

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

Note:

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

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

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

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

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

For more information on using network regions, with examples, see the application note *Network Regions for Avaya MultiVantage™ Solutions - A Tutorial*, which is available at: http://www.avaya.com/gcm/master-usa/en-us/resource/assets/applicationnotes/advantages_of_implem.pdf (requires Adobe Reader). For more information on configuring network regions in Avaya Communication Manager, see the application note *Avaya Communication Manager Network Region Configuration Guide*, which is available at: http://www.avaya.com/master-usa/en-us/resource/assets/applicationnotes/netw-region-tutorial.pdf (requires Adobe Reader). For information on using the Network Region Wizard, see the *Network Region Job Aid*, 14-300283, which is available at http://www.avaya.com/support.

Administering inter-network region connections

An **Alternate Routing Extension** field has been added to the second page of the **IP Network Region** screen. This unassigned extension (up to 7 digits long), together with two other fields are required for each network region in order to route the bearer portion of the IGAR call. The following must be performed:

- If IGAR is enabled for any row on pages 3 through 19, then the user shall be:
  - Required to enter an IGAR extension before submitting the screen
  - Blocked from blanking out a previously administered IGAR extension
- If IGAR is disabled by the System Parameter, the customer is warned if any of these fields are updated.

The warning is "WARNING: The IGAR System Parameter is disabled."

Type **`change ip-network-region #`** and press **Enter** to open the **Inter Network Region Connection Management** screen. Go to Page 2.

Figure 35: Alternate Routing Extension field

```
change ip-network-region 1                                    Page 2 of 19
                              IP NETWORK REGION

INTER-GATEWAY ALTERNATE ROUTING / DIAL PLAN TRANSPARENCY
 Incoming LDN Extension: 852-3999
 Conversion To Full Public Number - Delete: 0   Insert: +1732________
  Maximum Number of Trunks To Use for IGAR: 23
 Dial Plan Transparency in Survivable Mode? n

 BACKUP SERVERS(IN PRIORITY ORDER)          H.323 SECURITY PROFILES
 1   _________________                      1   challenge
 2   _________________                      2
 3   _________________                      3
 4   _________________                      4
 5   _________________
 6   _________________                      Allow SIP URI Conversion? y

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
   Near End Establishes TCP Signaling Socket? n
                        Near End TCP Port Min: 61440
                        Near End TCP Port Max: 61444
```

Pair-wise administration of IGAR between network regions

An **IGAR** column has been added to the **IP Network Region** screen to allow pair-wise configuration of IGAR between network regions. If the field is set to "y" the IGAR capability is enabled between the specific network region pair. If it is set to "n" the IGAR capability is disabled between the network region pair.

The following screen validations must be performed:

- If no IGAR Extension is administered on page 2 of the **IP Network Region** screen, the user is blocked from submitting the screen, if any network region pair has IGAR enabled.
- If IGAR is disabled using the System Parameter, the customer will be warned, if IGAR is enabled for any network region pair.

  The warning is "WARNING: The IGAR System Parameter is disabled."

Normally, the administration between Network Region pairs would have a codec set identified for compressing voice across the IP WAN. Only if bandwidth in the IP WAN is exceeded, and the **IGAR** field is set to "y", would the voice bearer be routed across an alternate trunk facility. However, under some conditions you may wish to force all calls to the PSTN.

The "forced" option can be used during initial installation to verify the alternative PSTN facility selected for a Network Region pair. This option may also be used to move traffic off of the IP WAN temporarily, if an edge router is having problems, or an edge router needs to be replaced between a Network Region pair.

When the codec set type is set to "pstn" the following fields are defaulted:

- **IGAR** field defaults to "y". Options: f(orced), n(o), y(es).

  This field must be defaulted to "y" because the Alternate Trunk Facility is the only means of routing the voice bearer portion of the call.

- When the codec set is set to "pstn" the following fields are hidden:
  - Direct-WAN
  - WAN-BW Limits, and
  - Intervening Regions

When the codec set is not "pstn" and not blank, the IGAR field is defaulted to "n".

A "f(orced)" option is supported in the **IGAR** column in addition to the options "n(o)" and "y(es)".

Figure 36: Inter network region connection management

```
change ip-network-region 3                                    Page 3 of 19

                    Inter Network Region Connection Management

src  dst   codec  direct   Audio           Video                              Dyn
rgn  rgn    set    WAN  WAN-BW Limits  WAN-BW Limits   Intervening-Regions   CAC  IGAR
3    1      1___   y    256:Kbits__    __________      ___ ___ ___ ___       ___   f
3    2      1___   n    ___________    __________      1__ ___ ___ ___       ___   y
3    3      1___   _    ___________    __________      ___ ___ ___ ___       ___   n
3    4      1___   n    ___________    __________      1__ ___ ___ ___       ___   n
3    5      1___   n    ___________    __________      6__ ___ ___ ___       ___   y
3    6      1___   _     __:NoLimit    __________      ___ ___ ___ ___       ___   y
3    7      1___   y    _10:Calls__    __________      ___ ___ ___ ___       ___   n
3    8      pstn   _    ___________    __________      ___ ___ ___ ___       ___   y
3    9      pstn   _    ___________    __________      ___ ___ ___ ___       ___   y
3    10
3    11
```

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

In the example, network region 3 is directly connected to regions 6, and 7, and is indirectly connected to regions 2 and 4 (through region 1) and 5 (through region 6).

Press **Enter** to save the changes.

Port network to network region mapping for boards other than IP boards

Existing IP Media Processor or Resource Modules, for example, the MedPro, C-LAN, and VAL, have assigned IP network regions. The new mapping from cabinet to IP Network Region does not override this administration.

The critical non-IP boards of interest are the trunk circuit packs over which IGAR calls are routed. When an IP connection between two port network/media gateways (PN/MGs) cannot be established, the system tries to establish an IGAR trunk connection between the two PN/MGs. The system tries to use trunks in the specific PN/MG requested. However, because Communication Manager does not require every PN/MG to have PSTN trunks, it may be necessary to obtain trunks from another PN/MG. The system may only obtain trunks from a PN/MG in the same Network Region as the one in which the original request was made. This means Communication Manager must let customers associate a port network with a Network Region. This can already be done with Media Gateways.

Note:

Cabinets connected through a center stage switch (CSS) are required to be in network region 1.

Figure 37: IP network region field on cabinet screen to map PNs to network regions

```
display cabinet 1                                                SPE B

                                  CABINET
 CABINET DESCRIPTION
                  Cabinet: 1
           Cabinet Layout: five-carrier
             Cabinet Type: processor
   Number of Portnetworks: 1
    Survivable Remote EPN? n
                 Location: 1_____________      IP Network Region: 1
        Cabinet Holdover: A-carrier-only
                     Room: 1K26______    Floor: _______  Building: 22_____

CARRIER DESCRIPTION
  Carrier       Carrier Type        Number      Duplicate

     C        port_____________     PN   01
     B        processor________     PN   01
     A        processor________     PN   01
     X        fan______________
     D        dup-sw-node______     SN   01        01E
     E        switch-node______     SN   01        01D
```

Status of inter-region usage

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

The IGAR Now/Today column on the **Inter Network Region Bandwidth Status** screen displays the number of times IGAR has been invoked for a network region pair, as shown in Figure 38. Type **status ip-network-region *n***, and press **Enter** to display the **Inter Network Region Bandwidth Status** screen.

Figure 38: IP network region status screen

```
status ip-network-region 2
                  Inter Network Region Bandwidth Status
                                                  Number of   # Times
Src Dst Conn     Conn  BW-Limit  BW-Used(Kbits) Connections  BW-Limit  IGAR
Rgn Rgn Type     Stat               Tx    Rx     Tx    Rx   Hit Today Now/Today
2   1   direct   pass  128 Kbits    xxx   xxx   xxx   xxx     xxx     xxx/ xxx
              Video:  NoLimit       xxx   xxx   xxx   xxx     xxx     xxx/ xxx
          Priority:  NoLimit       xxx   xxx   xxx   xxx     xxx     xxx/ xxx
2   3   indirect pass  NoLimit       xxx   xxx   xxx   xxx     xxx     xxx/ xxx
              Video:  NoLimit       xxx   xxx   xxx   xxx     xxx     xxx/ xxx
          Priority:  NoLimit       xxx   xxx   xxx   xxx     xxx     xxx/ xxx
2   4   indirect pass  NoLimit       xxx   xxx   xxx   xxx     xxx     xxx/ xxx
              Video:  NoLimit       xxx   xxx   xxx   xxx     xxx     xxx/ xxx
          Priority:  NoLimit       xxx   xxx   xxx   xxx     xxx     xxx/ xxx
2   11  indirect pass  NoLimit       xxx   xxx   xxx   xxx     xxx     xxx/ xxx
              Video:  NoLimit       xxx   xxx   xxx   xxx     xxx     xxx/ xxx
          Priority:  NoLimit       xxx   xxx   xxx   xxx     xxx     xxx/ xxx
```

The numbers in the column titled “IGAR Now/Today” have the following meanings:

- The first number (up to 3 digits or 999) displays the number of active IGAR connections for the pair of network regions at the time the command was invoked.
- The second number (up to 3 digits or 999) displays the number of times IGAR has been invoked for the pair of network regions since the previous midnight.

To administer the network region on the Signaling Group screen

Note:

The S8300 Media Server in LSP mode does not support signaling groups.

1. Type **change signaling-group *group#*** and press **Enter** to display the **Signaling Group** screen.

2. Type the number of the network region that corresponds to this signaling group in the **Far-end Network Region** field. The range of values is: **1-250** (S8300, S8500 or S8700-series servers)
3. Press **Enter** to save the changes.

Reviewing the network region administration

To check the network region administration:

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

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

                         ---- PHB Values ----  802.1p Priority     RSVP       Refr
Region    Name          Audio Video Ctrl BBE  Audio Video Ctrl   Profile     Rate
 1     Denver              46    26    34    46     0    5    7   guaranteed      15
 2     Cheyenne            19    19    19    46     0    2    1   controlled-load 15
```

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

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

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

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

Setting network performance thresholds

Note:

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

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

Note:

You cannot administer these parameters unless these conditions are met:

- The **Group Type** field on the **Signaling Group** screen is **h.323** or **sip**.
- 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 and are diverted to alternate routes.
  - Outgoing calls are blocked on this signaling group.

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

Note:

Avaya strongly recommends that you use the default values.

To administer network performance parameters

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

```
change system-parameters ip-options

                      IP-OPTIONS SYSTEM PARAMETERS

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

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

 AUTOMATIC TRACEROUTE ON
      Link Failure? n


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

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

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

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

Enabling spanning tree protocol (STP)

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

However, STP is slow to converge after a network failure, and slow to allow a new port into the network (~50 sec by default).

A modified version of STP, Rapid Spanning Tree converges faster than the earlier STP, and enables new ports much faster (sub-second) than the older protocol. **Rapid Spanning Tree** works with all Avaya equipment, and is *recommended* by Avaya.

To enable/disable spanning tree

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

   The full set of Spanning Tree commands is displayed in Figure 39.

Figure 39: Set Spantree commands

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

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

P330-1(super)#
```

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

   The selection of Spanning Tree protocol commands displays (see Figure 39).
5. To set the **rapid spanning tree** version, type **`set spantree version rapid-spanning-tree`** and press **Enter**.

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

Note:

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

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

Adjusting jitter buffers

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

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

Configuring UDP ports

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

About Media Encryption

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

What is Media Encryption?

To provide privacy for media streams that are carried over IP networks, Avaya Communication Manager supports encryption for IP bearer channel — voice data transported in Real Time Protocol (RTP) — between any combination of media gateways and IP endpoints.

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

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

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

What are the limitations of Media Encryption?

SECURITY ALERT:

Be sure that you understand these important media encryption limitations:

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

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

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

What types of media encryption are available?

Avaya Encryption Algorithm (AEA) and Advanced Encryption Standard (AES) are supported by most Avaya IP endpoints. Starting with Communication Manger release 4.0, the Secure Real Time Protocol (SRTP) encryption standard is supported by SIP endpoints and trunks and by the 9600-series telephones.

Table 18: Media Encryption support lists the telephones and Communication Manager releases that support each type of media encryption.

Table 18: Media Encryption support

	Media Encryption Type		
	AEA	**AES**	**SRTP**
Communication Manager release	CM 1.3 and later	CM 2.0 and later	CM 4.0 and later
Avaya IP telephones:			
4601	Y	Y	N
4602	Y	Y	N
4606	Y	N	N
4610SW	Y	Y	N
4612	Y	N	N
4620	Y	Y	N
4620SW / 4621SW / 4622SW / 4625SW / 4630SW	Y	Y	N
4624	Y	N	N
4630	Y	N	N
4690	N	N	N
9600-series IP telephones	Y	Y	Y
SIP endpoints	N	N	Y
IP Softphone	Y	Y	N
IP SoftConsole	Y	Y	N
IP Agent	Y	Y	N
TN2302AP IP Media Processor circuit pack	Y	Y	N
TN2602AP IP Media Resource 320 circuit pack	Y	Y	Y
VoIP elements of H.248 media gateways	Y	Y	Y

Is there a license file requirement?

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

Is Media Encryption currently enabled?

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

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

Media encryption field on Optional Features screen

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

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

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

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

Administering Media Encryption

This section contains Avaya Communication Manager administration procedures for:

- Administering Media Encryption for IP Codec Sets
- Administering Media Encryption for signaling groups

Note:

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

Administering Media Encryption for IP Codec Sets

The **IP Codec Set** screen enables you to administer the type of media encryption, if any, for each codec set.

Note:

See Table 15: IP Codec Set screen fields, page 1 on page 215 for a description of the fields on the IP Codec Set screen.

To administer media encryption on an IP codec set:

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

Media Encryption field on the IP Codec Set screen

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

                          IP Codec Set

    Codec Set: 7

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

Media Encryption:
1: 1-srtp-aescm128-hmac80
2: aes
3: aea
```

2. Enter up to three media encryption types listed in Table 19: Media Encryption Field Values (IP Codec Set) on page 251:

   Note:

   The option that you select for the **Media Encryption** field for each codec set applies to all codecs defined in that set.

   Note:

   This field is hidden if the **Media Encryption Over IP?** field on the **Customer Options** screen (Media encryption field on Optional Features screen on page 249) is *n*. The **Media Encryption** field appears only if the **Media Encryption over IP** feature is enabled in the license file (and displays as *y* on the **Customer Options** screen).

The **Media Encryption** field specifies one, two, or three options for the negotiation of encryption — in this example, one of the modes of **SRTP**, **aes**, and **aea**. You can specify no encryption by entering **none** in the **Media Encryption** field. The order in which the options are listed signifies the preference of use, similar to the list of codecs in a codec set. Two endpoints must support at least one common encryption option for a call to be completed between them.

The selected options for an IP codec set applies to all codecs defined in that set.

Table 19: Media Encryption Field Values (IP Codec Set)

Valid entries	Usage
aes	Advanced Encryption Standard (AES), a standard cryptographic algorithm for use by U.S. government organizations to protect sensitive (unclassified) information. AES reduces circuit-switched-to-IP call capacity by 25%.
aea	Avaya Encryption Algorithm. AEA is not as secure an algorithm as AES but call capacity reduction with AEA is negligible. Use this option as an alternative to AES encryption when: • All endpoints within a network region using this codec set must be encrypted. • All endpoints communicating between two network regions and administered to use this codec set must be encrypted.
SRTP — several encryption modes	SRTP provides encryption and authentication of RTP streams for calls between SIP-SIP endpoints, H.323-H.323 endpoints, and SIP-H.323 endpoints. SIP endpoints cannot use AEA or AES encryption. See Table 15: IP Codec Set screen fields, page 1 on page 215 for a list of SRTP encryption modes.
none	Media stream is unencrypted. This option prevents encryption when using this codec set and is the default setting when Media Encryption is not enabled.

Note:

The initial default value for this field is *none* when the **Media Encryption Over IP?** field in the **Optional Features** screen (on the **Customer Options** screen) is enabled (*y*) for the first time. If this field is *n*, the **Media Encryption** field on the **IP Codec Set** screen is hidden and functions as if *none* was selected.

Administering Media Encryption for signaling groups

To administer Media Encryption for an IP signaling group:

1. At the SAT type **change signaling-group *number*** to display the **Signaling Group** screen

Media encryption and passphrase fields for signaling groups

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

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

        Near-end Node Name:                   Far-end Node Name:
     Near-end Listen Port: 1720            Far-end Listen Port:
                                        Far-end Network Region:
             LRQ Required? n              Calls Share IP Signaling Connection? n
             RRQ Required? n
          Media Encryption? y                  Bypass If IP Threshold Exceeded? n
                Passphrase:                            H.235 Annex H Required? n
             DTMF over IP: out of band         Direct IP-IP Audio Connections? y
Link Loss Delay Timer(sec): 90                           IP Audio Hairpinning? n
                                              Interworking Message: PROGress
                                       DCP/Analog Bearer Capability: 3.1kHz
```

2. Enter ***y*** in the **Media Encryption?** field to enable Media Encryption on trunk calls using this signaling group.

Note:

Leaving this field in the default state (**n**) overrides the encryption administration on the IP Codec Set screen (Media Encryption field on the IP Codec Set screen on page 250) for any trunk call using this signaling group. That is, if the IP codec set that is used between two networks is administered as **aes** or **aea** (Table 19: Media Encryption Field Values (IP Codec Set) on page 251), then a call between two endpoints over a H.323 trunk using this IP codec set fails because there is no voice path.

This field does not display if the **Media Encryption Over IP?** field is `n` on the **Customer Options** screen (Media encryption field on Optional Features screen on page 249).

3. Type an 8- to 30-character string in the **Passphrase** field.

   This string:

   - Must contain at least 1 alphabetic and 1 numeric symbol
   - Can include letters, numerals, and!&*?;'^(),.:-
   - Is case-sensitive

   You must administer *the same passphrase* on both signaling group forms at each end of the IP trunk connection. For example, if you have two systems A and B with trunk A-B between them, you must administer both Signaling Group forms with *exactly the same passphrase* for the A-to-B trunk connection.

   If you have previously administered a passphrase, a single asterisk (*) appears in this field. If you have not administered a passphrase, the field is blank.

   Note:

   The **Passphrase** field does not appear if either the:

   - **Media Encryption Over IP?** field on the **Customer Options** screen (Media encryption field on Optional Features screen on page 249) is `n`.

     or

   - **Media Encryption?** field on the **Signaling Group** screen (Media encryption and passphrase fields for signaling groups on page 252) is `n`.

Viewing encryption status for stations and trunks

The current status of encryption usage by stations and trunks can be viewed using the **status station** and **status trunk** commands.

To check media encryption usage for a station, enter **status station <*extension*>**, and go to the **Connected Ports** page.

Connected ports screen

```
status station 60042                                    Page   6 of   7

                        SRC PORT TO DEST PORT TALKPATH
src port: s00001
S00001:TX:172.22.21.178:2976/g711u/20ms/1-srtp-aescm128-hmac80
S00001:TX:172.22.21.178:36226/g711u/20ms/1-srtp-aescm128-hmac80
```

This screen shows that a port is currently connected and using a G711 codec with SRTP media encryption.

To check media encryption usage for a trunk, enter **status trunk <*group/member*>**.

A display screen similar to the status station screen shows the trunk information.

About legal wiretapping

If you receive a court order requiring you to provide law enforcement access to certain calls placed to or from an IP endpoint, you can administer Service Observing permissions to a selected target endpoint (see Service Observing in Table 20: Media Encryption interactions on page 255). Place the observer and the target endpoint in a unique Class of Restriction (COR) with *exactly the same properties and calling permissions* as the original COR, otherwise the target user might be aware of the change.

About possible failure conditions

Using Media Encryption in combination with an administered security policy might lead to blocked calls or call reconfigurations because of restricted media capabilities. For example, if the IP codec set that is used between two network regions is administered as **aes** or **aea**, and if a call between two endpoints (one in each region) that do not support at least one common encryption option is set up, then there is no voice path.

How does Media Encryption interact with other features?

Media Encryption does not affect most Communication Manager features or adjuncts, except for those listed in Table 20: Media Encryption interactions on page 255.

Table 20: Media Encryption interactions

Interaction	Description
Service Observing	You can Service Observe a conversation between encrypted endpoints. The conversation remains encrypted to all outside parties except the communicants and the observer.
Voice Messaging	Any call from an encryption-enabled endpoint is decrypted before it is sent to a voice messaging system. When the TN2302AP IP Media Processor circuit pack receives the encrypted voice stream, it decrypts the packets before sending them to the voice messaging system, which then stores the packets in unencrypted mode.
Hairpinning	Hairpinning is not supported when one or both media streams are encrypted, and Avaya Communication Manager does not request hairpinning on these encrypted connections.
VPN	Media encryption complements virtual private network (VPN) security mechanisms. Encrypted voice packets can pass through VPN tunnels, essentially double-encrypting the conversation for the VPN “leg” of the call path.
H.323 trunks	Media Encryption behavior on a call varies based on these conditions at call set up: • Whether shuffled audio connections are permitted • Whether the call is an inter-region call • Whether IP trunk calling is encrypted or not • Whether the IP endpoint supports encryption • The media encryption setting for the affected IP codec sets These conditions also affect the codec set that is available for negotiation each time a call is set up. T.38 packets may be carried on an H.323 trunk that is encrypted; however the T.38 packet is sent in the clear.

About network management

Network management is the practice of using specialized software tools to monitor and maintain network components. Proper network management is a key component to the high availability of data networks.

The two basic network management models are:

- Distributed. Specialized, nonintegrated tools (and sometimes organizations) to manage discrete components
- Centralized. Integrating network management tools and organizations for a more coherent management strategy.

For a detailed discussion of Avaya's network management products, common third-party tools, and the distributed and centralized management models, see *Avaya Application Solutions: IP Telephony Deployment Guide* (555-245-600).

This section touches on the following topics:

- About H.248 link loss recovery
- Enterprise Survivable Servers (ESS)
- Controlling QoS policies
- Monitoring network performance

About H.248 link loss recovery

H.248 Link Loss Recovery is an automated way in which the media gateway reacquires the H.248 link when it is lost from either a primary call controller or an LSP. The H.248 link between a media server running Communication Manager and a media gateway, and the H.323 link between a media gateway and an H.323-compliant IP endpoint, provide the signaling protocol for:

- Call setup
- Call control (user actions such as Hold, Conference, or Transfer) while the call is in progress
- Call tear-down

If the link goes down, Link Recovery preserves any existing calls and attempts to re-establish the original link. If the gateway/endpoint cannot reconnect to the original server/gateway, then Link Recovery automatically attempts to connect with alternate TN799DP (C-LAN) circuit packs within the original server's configuration or to a Local Survivable Processor (LSP).

Overlap with the Auto Fallback to Primary feature occurs when the Link Loss Recovery starts while the media gateway is trying to migrate back to the primary, with its new registration message indicating that service is being obtained from elsewhere.

A race condition may exist in which there is an outstanding media gateway registration to the primary while the link to the LSP is lost. The media gateway awaits a denial or acceptance from the primary call controller. If it is an acceptance, then the Link Loss Recovery is terminated, and the media gateway is serviced by the primary call controller. If it is a denial, then the media gateway immediately sends a new registration to the primary call controller indicating no service, and the existing H.248 Link Loss Recovery feature takes over.

These features are similar in that they both attempt to return service to the primary call controller; however, Link Loss Recovery does it based upon a link failure, whereas auto fallback to primary does it based upon a working fragmented network.

Auto fallback to primary controller for H.248 media gateways

The intent of the auto fallback to primary controller feature is to return a fragmented network, in which a number of H.248 Media Gateways are being serviced by one or more LSPs (Local Survivable Processors), to the primary media server in an automatic fashion. This feature is targeted towards all H.248 media gateways. By migrating the media gateways back to the primary automatically, the distributed telephony switch network can be made whole sooner without human intervention, which is required today.

The auto-fallback migration, in combination with the connection preservation feature for H.248 gateways is connection-preserving. Stable connections are preserved; unstable connections (such as ringing calls) are not. There still may be a very short interval without dialtone for new calls.

The media gateway presents a new registration parameter that indicates that Service is being obtained from an LSP, and indicates the number of active user calls on the media gateway platform. The server administers each media gateway to have its own set of rules for Time of Day migration, enable/disable, and the setting of call threshold rules for migration.

This feature allows the administrator to define any of the following rules for migration:

- The media gateway should migrate to the primary automatically, or not.
- The media gateway should migrate immediately when possible, regardless of active call count.
- The media gateway should only migrate if the active call count is 0.
- The media gateway should only be allowed to migrate within a window of opportunity, by providing day of the week and time intervals per day. This option does not take call count into consideration.
- The media gateway should be migrated within a window of opportunity by providing day of the week and time of day, *or immediately* if the call count reaches 0. Both rules are active at the same time.

Internally, the primary call controller gives priority to registration requests from those media gateways that are currently not being serviced by an LSP. This priority is not administrable.

There are several reasons for denying an auto-fallback, which can result from general system performance requirements, or from administrator-imposed requirements. General system performance requirements can include denial of registration because:

- Too many simultaneous media gateway registration requests

Administrator-imposed requirements for denial of a registration can include:

- Registrations restricted to a windowed time of day
- Migration restricted to a condition of 0 active calls, that is, there are no users on calls within the media gateway in question.
- The administered minimum time for network stability has not been exceeded.

Other characteristics of this feature include:

- This feature does not preclude an older GW firmware release from working with Communication Manager 3.0 or vice versa; however, the auto-fallback feature would not be available.

  For this feature to work, the call controller is required to have Communication Manager 3.0, while the media gateway is required to have the GW firmware available at the time of the Communication Manager 3.0 release.
- Existing H.248 media gateways are the targets.
- LSP operation is completely unaffected.

  The LSP simply sees that a particular media gateway has lost its connection with the LSP. The existing H.248 Link Loss Recovery algorithm on the LSP cleans up all outstanding call records within the LSP after the prescribed time interval.

Basic feature operation

The following steps illustrate the basic operation of the auto-fallback to primary for H.248 media gateways feature. While not exactly so, the steps are approximately sequential.

1. The media gateway/media server *by default* has this feature disabled.

   If the media gateway is initially registered with an older media server, the version information exchange is sufficient for the media gateway to know not to attempt to fallback to the primary automatically.

2. By means of administration on the media server, this feature can be enabled for any or all media gateways controlled by that media server.

   The *enable/disable* administration on the media server determines whether the media server will *accept/deny* registration requests containing the new parameter that service is being obtained from an LSP. The media gateway continuously attempts to register with the media server, however, even if the media server has been administered never to accept the registration request (that is, the auto-fallback feature is disabled on the media server). In such a case, a manual return of the media gateway is required, which generates a different registration message that is accepted by the media server.

Note:

There is still value in receiving the registration messages when auto-fallback is disabled on the media server, and that value is to see the stability of the network over time, since those messages act as "keep-alive" messages.

3. The permission-based rules that include time of day and context information are only known to the media server.

   There is no need for the LSP to have any of these translations.

4. When associated with a primary controller running Communication Manager 3.0, the media gateway attempts to register with the primary controller whenever it is connected to an LSP.

   This registration attempt happens every 30 seconds, once the media gateway is able to communicate with the primary controller. The registration message contains an element that indicates:

   - that the media gateway is being serviced by an LSP, and
   - the number of active user calls on that media gateway.

5. Upon the initial registration request, the primary controller initializes the encrypted TCP link for H.248 messaging.

   This is performed regardless of whether that initial registration is honored or not, and that encryption is maintained throughout the life of the registration requests. The encryption is also maintained once a registration is accepted by the primary controller. Encryption of the signaling link is performed at the outset during this automatic fallback process to ensure the security of the communication between the primary call controller and the media gateway.

6. The primary controller, based upon its administered rules, may allow or deny a registration.

   If the primary controller gets a registration message without Service State information, for example, an older media gateway, or if a new media gateway states it does not have service, then the primary honors those registration requests above all others immediately.

7. If the registration is denied, the media gateway continues to send the registration message every 30 seconds, which acts as a *de facto* '"keep-alive" message.

8. The media gateway constantly monitors the call count on its platform, and asynchronously sends a registration message whenever 0 context is achieved.

9. Once the registration message is accepted by the primary, then the H.248 link to the LSP is dropped.

G250 interworking

When calls are made on the media gateway while it is controlled by Standard Local Survivability (SLS), the G250 behaves as any LSP might behave. The SLS, using its administration and dial analysis plan, can allow local calls to be established from:

- Local station to local station (analog or registered IP)
- Local station to local analog two-way CO trunks

While operating in SLS mode, the G250 attempts to re-register with the primary controller on its MGC list. As soon as the gateway is able to re-register with the primary controller, it un-registers with SLS, and re-registers with the primary controller. In terms of re-registration with the primary controller, the Auto Fallback to Primary feature would therefore work in a similar way with the G250 SLS as it does with the LSPs in the G350 or G700.

Note:

The connection preserving aspects of this feature will not be available on the G250 for this release.

G350 interworking

The G350 firmware loads use the Object Identifier (OID) that has the longer Non-Standard Data format in the registration message. This format is only backward compatible to Communication Manager 2.0 loads. Older loads respond with a protocol error as the denial cause for the rejection of the new registration message. Given that the G350 was only introduced in the Communication Manager 2.0 timeframe, it is not backwards compatible with previous Communication Manager releases.

In a startup scenario, there is an exchange of version information between Communication Manager and the media gateway. If the Communication Manager load is pre-Communication Manager 3.0, then the auto-fallback mechanism remains disabled for the media gateway. Any subsequent registration with a primary controller (from the MGC list) that is running release Communication Manager 3.0 results in the auto fall-back feature being enabled for the media gateway.

The only time when the media gateway may send a registration message to an older primary call controller is in the rare case when the primary controller has been downgraded while the media gateway has been receiving service from an LSP. In this case, the media gateway receives a protocol error that can be used to send a registration message consistent with Communication Manager 2.0. Downgrading to earlier than Communication Manager 2.0 with a G350 would result in the G350 not being able to register at all.

G700 interworking

The G700 Media Gateway, even in Communication Manager 2.0, still used the same OID as when it was originally deployed. The OID available for the G350 was not ported to the G700. The auto fallback to primary feature requires that all G700s, running the Communication Manager 3.0-compliant firmware load, use the OID format. The NSD (Non-Standard Data) expansion with the OID is used to carry the context count.

If the media gateway receives any of the following errors in response to a registration message, then the media gateway sends the original OID registration message prior to the expansion of the NSD.

- 284 - NSD OID invalid
- 283 - NSD OID wrong length
- 345 - NSD Wrong Length - for Communication Manager 1.3 and earlier systems

Though not directly necessary for this feature, the media gateway responds to any of the aforementioned protocol errors by attempting to register with the lowest common denominator registration message. This allows new media gateways to be backward compatible with even older releases. This modification only applies to the G700.

Older media gateway loads

The auto-fallback feature on the media server is passive in nature; therefore, an older media gateway load trying to register with the new Communication Manager 3.0 load registers with priority, since the value of the Service-State is that of a media gateway without service. Any defined rules for the media gateway are ignored, given that an older media gateway firmware release tries to register only when it no longer has service from another media server; therefore, the administration of rules for old media gateway firmware loads are irrelevant.

Administering auto fallback to primary

For each media gateway, the following administration must be performed:

- Adding Recovery Rule to Media Gateway screen
- Administering the System Parameters Media Gateway Automatic Recovery Rule screens to schedule the auto-fallback within the system-parameters area.

Adding Recovery Rule to Media Gateway screen

The **Media Gateway** screen (**`change media-gateway n`**) has a field called **Recovery Rule** with the following attributes:

- Acceptable values for the field are **none**, **1 - 50**, or **1 - 250**, where
  - **50** is the maximum number of supported media gateways on an S8300 Media Server, and
  - **250** is the maximum number of supported media gateways on an S8500 or S8700-series Media Server.
- Default is **none**, which indicates that no automatic fallback registrations will be accepted.
- The value of **1 - 50**, or **1 - 250** applies a specific recovery rule to that numbered gateway.

  Note:

  A single recovery rule number may be applied to all media gateways, or each media gateway may have its own recovery rule number, or any combination in between.

By associating the recovery rule to the **Media Gateway** screen (see Figure 40), an administrator can use the **`list media-gateway`** command to see which media gateways have the same recovery rules. All the administration parameters for the media gateways are consolidated on a single screen. The actual logic of the recovery rule is separate, but an administrator can start from the Media Gateway screen and proceed to find the recovery rule.

Note:

These changes apply to the **`display media-gateway`** command, as well.

Figure 40: Media Gateway screen

```
change media-gateway 1                                        Page 1 of 1
                                MEDIA GATEWAY

         Number: 1                            IP Address: xxx.xxx.xxx.xxx
           Type: g350              Fw Version/HW Vintage: xxx.yyy.zzz/nnn
                                      MAC Address: 00:04:0d:00:00:64
  Serial Number:                     Encrypt Link? y
 Network Region: 1                       Location: 1
    Registered? y                  Controller IP Address: xxx.xxx.xxx.xxx
 Recovery Rule: none                        Site Data:
     Slot    Module Type              Name
      V1:      S8300                 ICC MM
      V2:      MM714                 4+4 ANA MM
      V3:      MM722                 2 TRUNK BRI MM
      V4:      MM710                 DS1 MM


      V8:
      V9:
```

In the above example, no automatic fallback registration requests will be accepted by the primary controller for Media Gateway 1 when it is active on an LSP.

Note:

For more detailed descriptions of the entries and values fields on this screen, see *Maintenance Commands for Avaya Communication Manager, Media Gateways and Servers*, 03-300431, at http://www.avaya.com/support).

Administering the System Parameters Media Gateway Automatic Recovery Rule screens

Definition of recovery rules occurs on the **System Parameters Media Gateway Automatic Recovery Rule** screens (**`change system-parameters mg-recovery-rule <n>`**. This screen is contained within the 'system-parameters' area of administration screens. The maximum number of screens that can be administered correspond to the maximum number of media gateways supported by the media server in question, and are:

- Up to 50 for the S8300 Media Server
- Up to 250 for the S8500 and S8700-series Media Servers

These screens provide a field, **Migrate H248 MG to primary**, with 4 administrable options:

Note:

For detailed information on all four options, see *Administrator Guide for Avaya Communication Manager,* 03-300509.

1. **immediately** — which means that the first media gateway registration that comes from the media gateway is honored, regardless of context count or time of day.

   The Warning displayed in Figure 41 is visible when a user selects this option. This option is the default for all rules.

2. **0-active calls** — which means that the first media gateway registration reporting “0 active calls” is honored (see Figure 42).

3. **Time-day-window** — means that a valid registration message received during any part of this interval is honored (see Figure 43).

   Note:

   Time of day is local to the media gateway.

   There are no constraints on the number of active calls. The time scale provided for each day of the week goes from 00-23 hundred hours (military time). The user must specify an ‘x’ or ‘X’ for each hour where they want to permit the return migration. If they do not want to permit a given hour, then they leave it blank. This method gets around overlapping time issues between days of the week. Users can specify as many intervals as they wish.

4. **Time-window-OR-0-active-calls** — means that a valid registration is accepted *anytime,* when a 0 active call count is reported OR if a valid registration with *any* call count is received during the specified time/day intervals (see Figure 44).

   Note:

   Time of day is local to the media gateway.

   The time scale provided for each day of the week goes from 00-23 hundred hours (military time). The user must specify an ‘x’ or ‘X’ for each hour where they want to permit the return migration. If they do not want to permit a given hour then they leave it blank. This method gets around overlapping time issues between days of the week. Users can specify as many intervals as they wish.

Figure 41: System-parameters mg-recovery-rule screen: immediately

```
change system-parameters mg-recovery-rule <n>

         SYSTEM PARAMETERS MEDIA GATEWAY AUTOMATIC RECOVERY RULE

Recovery Rule Number: n
Rule Name:
Migrate H.248 MG to primary:   immediately
Minimum time of network stability: 3

WARNING: The MG shall be migrated at the first possible opportunity. The MG
may be migrated with a number of active calls. These calls shall have their
talk paths preserved, but no additional call processing of features shall be
honored. The user must hang up in order to regain access to all features.


Note: set 'Migrate H.248 MG to primary' to Blank to disable rule.
```

Administer the following fields:

Field	Description
Recovery Rule Number	The number of the recovery rule: • Up to 50 for the S8300 Media Server • Up to 250 for the S8500 and S8700-series Media Servers
Rule Name	Optional text name for the rule, to aid in associating rules with media gateways.
Migrate H.248 MG to primary	One of 4 administrable options.
Minimum time of network stability	Administrable time interval for stability in the H.248 link before auto-fallback is allowed. Between 3-15 minutes (Default is 3 minutes).

Figure 42 shows the screen for the **0-active calls** option.

Figure 42: System-parameters mg-recovery-rule screen: 0-active calls

```
change system-parameters mg-recovery-rule <n>

         SYSTEM PARAMETERS MEDIA GATEWAY AUTOMATIC RECOVERY RULE

Recovery Rule Number: n
Rule Name:
Migrate H.248 MG to primary: __0-active-calls___
Minimum time of network stability: 3

WARNING: The MG shall only be migrated when there are no active calls.




Note: set 'Migrate H.248 MG to primary' to Blank to disable rule.
```

Figure 43 shows the screen for the time-day-window option.

Figure 43: System-parameters mg-recovery-rule screen: time-day-window

```
change system-parameters mg-recovery-rule n
         SYSTEM PARAMETERS MEDIA GATEWAY AUTOMATIC RECOVERY RULE
Recovery Rule Number: n
Rule Name:
Migrate H.248 MG to primary: __time-day-window___
Minimum time of network stability: 3
WARNING:  The MG may be migrated with a number of active calls.  These calls
shall have their talk paths preserved, but no additional call processing of
features shall be honored.  The user must hang up in order to regain access
to all features.  Valid registrations shall only be accepted during these
intervals.
                        Time of Day
             00                        12                       23
Day of week
Sunday        _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _ _  _
Monday        _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _ _  _
Tuesday       _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _ _  _
Wednesday     _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _ _  _
Thursday      _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _ _  _
Friday        _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _ _  _
Saturday      _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _ _  _

Note: set 'Migrate H.248 MG to primary' to Blank to disable rule.
```

Figure 44 shows the screen for the **time-window-OR-0-active-calls** option.

Figure 44: System-parameters mg-recovery-rule screen: time-window-OR-0-active-calls

```
change system-parameters mg-recovery-rule n

         SYSTEM PARAMETERS MEDIA GATEWAY AUTOMATIC RECOVERY RULE

Recovery Rule Number: 1
Rule Name:
Migrate H.248 MG to primary: __time-window-OR-0-active-calls___
Minimum time of network stability: 3
WARNING:  The MG shall be migrated at ANY time when there are no active
calls, OR the MG may be migrated with a number of active calls when a
registration is received during the specified intervals below.  These calls
shall have their talk paths preserved, but no additional call processing of
features shall be honored.
                          Time of Day
             00                          12                          23
Day of week
Sunday        _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _  _
Monday        _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _  _
Tuesday       _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _  _
Wednesday     _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _  _
Thursday      _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _  _
Friday        _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _  _
Saturday      _  _ _ _ _ _ _ _ _ _ _ _  _  _ _ _ _ _ _ _ _ _ _  _

Note: set 'Migrate H.248 MG to primary' to Blank to disable rule.
```

For administrators to see how the recovery rules are applied across all media gateways, the **Media Gateway Report** screen (**`list media-gateway`** command) identifies the recovery rule for each media gateway in the network (See Figure 45).

Figure 45: list mg-recovery screen

```
list media-gateway                                          Page 1 of 1
                          MEDIA GATEWAY REPORT

Num  Name                 Serial No/        IP Address/     Type  NetRgn/   Reg?
                          FW Ver/HW Vint    Cntrl IP Addr         RecRule

1    GW#1 Boxster Lab     01DR11131345      135.8  .77 .62  g700   1        n
                          unavailable                               none

2    MG2 Boxster MV Lab   02DR06750093                       g700   1        n
                          unavailable                               10

3    MG3 Boxster MV Lab   01DR10245104      135.8  .77 .68  g700   1        n
                          unavailable                               none
```

In this example, media gateways #1 and #3 are administered such that no registration request would be accepted by the primary controller when the media gateway is active on an LSP. Media gateway #2, on the other hand, is administered with Recovery Rule #10. The SAT command:

display system-parameters mg-recovery-rule ***10***

would show the details of that specific recovery rule.

Enterprise Survivable Servers (ESS)

The Enterprise Survivable Servers (ESS) feature provides survivability to port networks by allowing backup servers to be placed in various locations in the customer's network. The backup servers supply service to port networks in the case where the S8500 media server, or the S8700-series media server pair fails, or connectivity to the main Communication Manager server(s) is lost. ESS servers can be either S8500 or S8700-series Media Servers; an S8500 can back up an S8500 or S8700, and an S8700-series server can also be used to back up a corresponding S8700-series server. ESS servers offer full Avaya Communication Manager functionality when in survivable mode, provided sufficient connectivity exists to other Avaya components (for example, endpoints, gateways, and messaging servers). One exception is that an ESS cannot control a Center Stage Switch.

When designing a network to support ESS servers, consider the following:

- ESS servers can only control port networks that they can reach over an IP-connected or ATM-connected network.

  That is, ESS servers connected on an enterprise's public IP network will not be able to control port networks connected to control network A or B, unless:

  - ESS can control a remote port network that is connected through ATM to port networks on control networks A or B, or
  - Control networks A or B are exposed to the public IP network through control network on the Customer's LAN (CNOCL).

- Multiple ESSs can be deployed in a network. In the case above, an enterprise could deploy one or more ESSs on the public network, and an additional server on control networks A and B to backup port networks attached to the respective networks.

  However, when port networks register with different ESS servers, system fragmentation may occur. In that case, care should be taken to establish adequate routing patterns to allow users at a particular location to be able to place calls where needed.

- ESS servers register to the main server(s) through a C-LAN. Each ESS must be able to communicate with a C-LAN in order to download translations from the main server. The file synchronization process uses the following ports:
  - UDP/1719 – ESS registers with the main server
  - TCP/21873 – Main server sends translations to the LSP(s) (pre-Release 3.0)
  - TCP/21874 – Main server sends translations to the ESS (Release 3.0 and above; also for LSP translations)

The media gateway cannot distinguish between registration through a C-LAN or registration to an S8300 directly. Prior to Communication Manager 3.0, without ESS, if a media gateway successfully registered with a primary call controller IP address, then the media gateway was properly registered with the primary call controller. However, in Communication Manager 3.0, when a media gateway completes a successful registration through an IP address defined as a primary call controller address, if that address is a C-LAN, the media gateway may not necessarily be registered with the true primary call controller. The port network that houses the C-LAN may be under control of an ESS, but the media gateway will not know that it is registered with an ESS.

When the traditional port network migrates back to the primary call controller, then the media gateway loses its H.248 link, and the Link Loss Recovery algorithm engages, and that should be sufficient. The Auto Fallback to Primary feature only engages if the media gateway drops the connection and registers with an LSP. The ESS migration should only occur if the port network is reasonably certain to return to the primary call controller, so the media gateway would simply return to the same C-LAN interface. Now, when the media gateway returns to the same C-LAN interface, the Link Loss Recovery feature performs a context audit with the primary controller and learns that the primary call controller is not aware of the media gateway. The controller in this case issues a warm start request to the media gateway, or potentially different behavior if connection preservation is active at the same time. The auto-fallback feature is not affected by ESS.

For more information on ESS, see the *Using the Avaya Enterprise Survivable Servers (ESS)*, 03-300428.

Controlling QoS policies

Avaya Policy Manager is a network management tool that allows you to control Quality of Service (QoS) policies in your IP voice network consistently:

- Avaya Policy Manager helps you implement QoS policies consistently for both the data and the voice networks.
- QoS policies are assigned according to network regions and are distributed through the Enterprise Directory Gateway to your systems and to routers and switching devices.

Figure 46: Avaya Policy Manager application sequence on page 269 illustrates how Avaya Policy Manager works.

Figure 46: Avaya Policy Manager application sequence

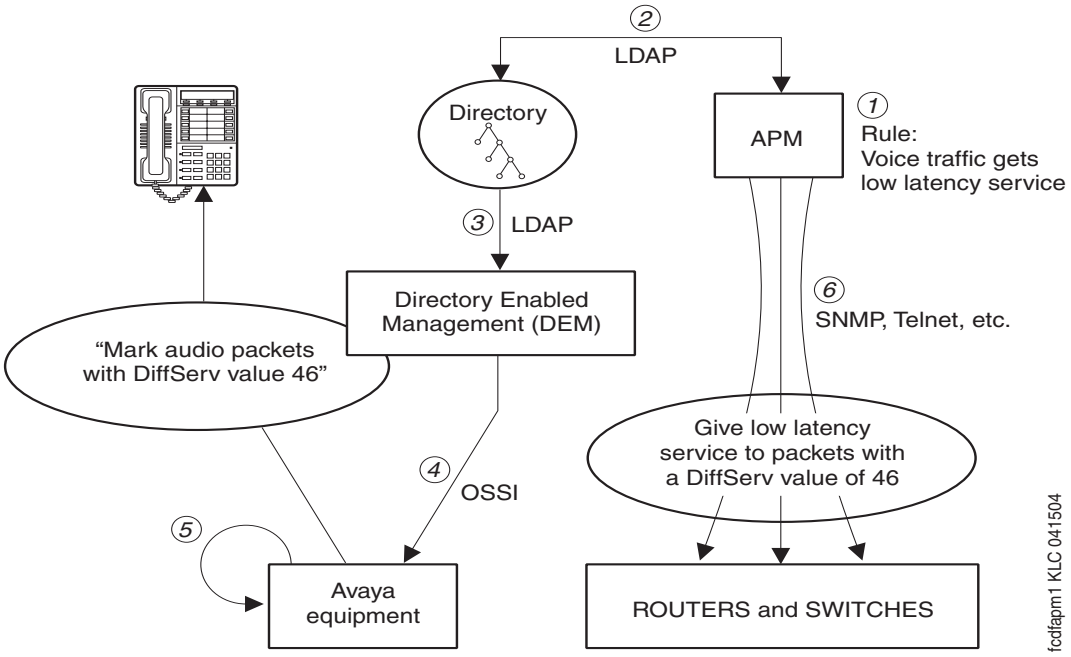


Figure notes:

1. **Business rule established in Avaya Policy Manager**
2. **Avaya Policy Manager uses LDAP to update Communication Manager**
3. **Directory Enabled Management (DEM) identifies the change in the directory.**
4. **EDG updates Communication Manager administration through the Ethernet switch**
5. **Communication Manager tells the Media Processor, C-LAN, and IP Phones to mark audio packets with DSCP=46.**
6. **Avaya Policy Manager distributes policy information to other network devices, including low latency service for DiffServ value of 46.**

For more information about Avaya Policy Manager, see your Avaya representative.

Monitoring network performance

The Avaya VoIP Monitoring Manager, a VoIP Network Quality monitoring tool, allows you to monitor these quality-affecting network factors:

- Jitter levels
- Packet loss
- Delay
- CODECs used
- RSVP status

For more information about Avaya VoIP Monitoring Manager, see *Avaya Application Solutions: IP Telephony Deployment Guide* (555-245-600).

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