Nortel Communication Server 1000

IP Peer Networking Installation and Commissioning

NN43001-313

Document status: Standard
Document version: 01.01
Document date: 30 May 2007

Sourced in Canada.

The information in this document is subject to change without notice. The statements, configurations, technical data, and recommendations in this document are believed to be accurate and reliable, but are presented without express or implied warranty. Users must take full responsibility for their applications of any products specified in this document. The information in this document is proprietary to Nortel Networks.

Revision history

May 2007

Standard 01.01. This document is a new NTP for Communication Server 1000 Release 5.0. It was created to support a restructuring of the Documentation Library. This document is comprised of (1) information on IP Peer Networking that was previously contained in the legacy document, now retired: *IP Peer Networking: Installation and Configuration* (553-3001-213), and (2) a description of the installation and configuration of Communication Server 1000 Release 5.0 IP Peer Networking.

April 2007

Standard 10.00. This document is up-issued for: (1) CR Q01454475, revising the configuration rules for Bandwidth Management. See page 140. (2) CR Q01524156, revising the description of loop limitations on a large system. See page 303. (3) CR Q0158368, revising the default value of the FOPT (Flexible Orbit Prevention Timer) from 14 to 6 seconds. See page 363. (4) CR Q01524220, specifying that a user password can be up to 24 characters in length. See page 480.

December 2006

Standard 9.00. This document is up-issued for CR Q01453520, specifying that the Primary, Alternate and (optional) Failsafe Network Routing Servers must host the same major software release. See page 384.

November 2006

Standard 8.00. This document is up-issued for CR Q014694590-01, specifying that at least 768 MByte of memory is required on the Signaling Server to obtain 1200 H.323 Virtual Trunks. See Table 1: “Virtual Trunk limits for each Signaling Server” on page 29.

October 2006

Standard 7.00. This document is up-issued for CR Q01461442, specifying in the Procedure for Adding a Collaborative Server that the TLAN IP address of the server must be entered in the Server address text box. See page 445.

August 2006

Standard 6.00. This document is up-issued for CR Q01374118, adding a note that Data calls are not supported on Virtual Trunks. See page 28.

April 2006

Standard 5.00. This document is upissued for CR Q01256567-01, adding a statement that Nortel does not support a modem in IP networks. See pages 27 and 182.

January 2006

Standard 4.00. This document is up-issued for CR Q01202736, with information on reconfiguring Call Server alarm notification levels if necessary when configuring Adaptive Network Bandwidth Management. See pages 158 and 166.

August 2005

Standard 3.00. This document is up-issued to support Communication Server 1000 Release 4.5.

September 2004

Standard 2.00. This document is up-issued for Communication Server 1000 Release 4.0.

October 2003

Standard 1.00. This document is a new NTP for Succession 3.0. It was created to support a restructuring of the Documentation Library. This document contains information previously contained in the following legacy document, now retired: *IP Peer Networking (553-3023-220)*.

Contents

Procedures

New in this Release

WARNING
Do ***not*** contact Red Hat for technical support on your Nortel version of the Linux base operating system. If technical support is required for the Nortel version of the Linux base operating system, contact Nortel technical support through your regular channels.

Communication Server 1000 Release 5.0 introduces the following IP Peer Networking features:

- Linux-based Network Routing Service (NRS) with SIP Proxy and Redirect Server in Active/Active database mode
- Linux-based Network Routing Service Manager (NRS Manager) as a component of Enterprise Common Manager (ECM)
- Transport Layer Security (TLS) and Transport Layer Port normalization for SIP
- SIP INVITE URI Network Post-Translation
- RFC 2833 for SIP VTRK calls
- VTRK GW TLAN Health Monitoring
- ECM Private CA managing Public Key Certificates on Signaling Servers for SIP/TLS and HTTPS
- Network-wide RAN and MOH
- Interoperation with MCM for Microsoft LCS 2005/OCS 2007

How to get help

This chapter explains how to get help for Nortel products and services.

Finding the latest updates on the Nortel Web site

The content of this documentation is current at the time the product is released. To check for updates to the latest documentation for Communication Server (CS) 1000, go to www.nortel.com and navigate to the Technical Documentation page for CS 1000.

Getting help from the Nortel Web site

The best way to get technical support for Nortel products is from the Nortel Technical Support Web site:
www.nortel.com/support

This site provides quick access to software, documentation, bulletins, and tools to address issues with Nortel products. From this site, you can:

- download software, documentation, and product bulletins
- search the Technical Support Web site and the Nortel Knowledge Base for answers to technical issues
- sign up for automatic notification of new software and documentation for Nortel equipment
- open and manage technical support cases

Getting help over the telephone from a Nortel Solutions Center

If you do not find the information you require on the Nortel Technical Support Web site, and you have a Nortel support contract, you can also get help over the telephone from a Nortel Solutions Center.

In North America, call 1-800-4NORTEL (1-800-466-7835). Outside North America, go to the following Web site to obtain the telephone number for your region:
www.nortel.com/callus

Getting help from a specialist by using an Express Routing Code

To access some Nortel Technical Solutions Centers, you can use an Express Routing Code (ERC) to quickly route your call to a specialist in your Nortel product or service. To locate the ERC for your product or service, go to: www.nortel.com/erc

Getting help through a Nortel distributor or reseller

If you purchased a service contract for your Nortel product from a distributor or authorized reseller, contact the technical support staff for that distributor or reseller.

Introduction

This document is a global document. Contact your system supplier or your Nortel representative to verify that the hardware and software described are supported in your area.

Subject

This document describes the IP Peer Networking feature, and how to implement IP Peer Networking as part of your system.

Note on legacy products and releases

This NTP contains information about systems, components, and features that are compatible with Nortel Communication Server 1000 Release 5.0 software. For more information on legacy products and releases, click the **Technical Documentation** link under **Support & Training** on the Nortel home page:

http://www.nortel.com

Applicable systems

This document applies to the following systems:

- Communication Server 1000M Single Group (CS 1000M SG)
- Communication Server 1000M Multi Group (CS 1000M MG)
- Communication Server 1000E (CS 1000E)
- Meridian 1 PBX 11C Chassis
- Meridian 1 PBX 11C Cabinet
- Meridian 1 PBX 61C
- Meridian 1 PBX 81C

> ***Note:*** When upgrading software, memory upgrades may be required on the Signaling Server, the Call Server, or both.

Intended audience

This document is intended for administrators responsible for configuring the IP Peer Networking feature and managing the Network Routing Service database.

Conventions

Terminology

In this document, the following systems are referred to generically as *system*:

- Communication Server 1000M (CS 1000M)
- Communication Server 1000E (CS 1000E)
- Meridian 1

The following systems are referred to generically as *Small System*:

- Meridian 1 PBX 11C Chassis
- Meridian 1 PBX 11C Cabinet

The following systems are referred to generically as *Large System*:

- Communication Server 1000M Single Group (CS 1000M SG)
- Communication Server 1000M Multi Group (CS 1000M MG)
- Communication Server 1000E (CS 1000E)
- Meridian 1 PBX 61C
- Meridian 1 PBX 81C

Unless specifically stated otherwise, the term *Element Manager* refers to the CS 1000 Element Manager.

Related information

This section lists information sources that relate to this document.

NTPs

The following NTPs are referenced in this document:

- *Converging the Data Network with VoIP* (NN43001-260)
- *Electronic Switched Network: Signaling and Transmission Guidelines* (NN43001-280)
- *Dialing Plans: Description* (NN43001-283)
- *Signaling Server Installation and Commissioning* (NN43001-312)
- *Branch Office Installation and Commissioning* (NN43001-314)

- *System Management* (NN43001-600)
- *Features and Services* (NN43001-106)
- *System Redundancy Fundamentals (NN43001-507)* (NN43001-507)
- *Software Input/Output: Administration* (NN43001-611)
- *Telephony Manager 3.1: System Administration* (NN43050-601)
- *IP Trunk: Description, Installation, and Operation* (NN43001-563)
- *IP Line: Description, Installation, and Operation* (NN43100-500)
- *Basic Network Features* (NN43001-579)
- *Software Input/Output: System Messages* (NN43001-712)
- *Software Input/Output: Maintenance* (NN43001-711)
- *Simple Network Management Protocol: Description and Maintenance* (NN43001-719)
- *Communication Server 1000M and Meridian 1: Small System Planning and Engineering* (NN43011-220)
- *Communication Server 1000M and Meridian 1: Small System Installation and Commissioning* (NN43011-310)
- *Communication Server 1000M and Meridian 1: Small System Upgrade Procedures* (NN43011-459)
- *Communication Server 1000M and Meridian 1: Large System Planning and Engineering* (NN43021-220)
- *Communication Server 1000M and Meridian 1: Large System Installation and Commissioning* (NN43021-310)
- *Communication Server 1000M and Meridian 1: Large System Upgrade Procedures* (NN43011-459)
- *Communication Server 1000E: Planning and Engineering (NN43041-220)* (NN43041-220)
- *Communication Server 1000E: Installation and Commissioning* (NN43041-310)
- *Communication Server 1000E: Upgrade Procedures* (NN43041-458)
- *Communication Server 1000E: Maintenance* (NN43041-700)
- *Multimedia Portfolio Communication (MCP) Interworking Basics* (NN42020-127)
- *CallPilot Planning and Engineering Guide (NN44200-200)* (553-7101-101)
- *CallPilot Installation and Configuration Part 3: T1/SMDI and CallPilot Server Configuration (NN44200-303)* (553-7101-224)

- *CallPilot Administrator’s Guide (44200-601)* (553-7101-301)

Online

To access Nortel documentation online, click the **Technical Documentation** link under **Support & Training** on the Nortel home page: www.nortel.com

CD-ROM

To obtain Nortel documentation on CD-ROM, contact your Nortel customer representative.

Overview

Contents

This section contains information on the following topics:

IP Peer Networking overview

WARNING
Do ***not*** contact Red Hat for technical support on your Nortel version of the Linux base operating system. If technical support is required for the Nortel version of the Linux base operating system, contact Nortel technical support through your regular channels.

IP Peer Networking enables the customer to distribute the functionality of the CS 1000 systems over a Wide Area Network (WAN), using either Nortel Session Initiation Protocol (SIP) or H.323 Gateways or other third-party SIP or H.323 Gateways.

Key advantages of IP Peer Networking are as follows:

- Provides global coverage of standard Voice over Internet Protocol (VoIP) signaling interfaces.
- Enables the networking of multiple systems across an IP network.
- Enables the customer to provision IP Phones anywhere on the connected network (LAN/MAN/WAN) and also enables them to provide LAN-connected modules (such as a router, Layer 2 switch, Layer 3 switch, bridge, or hub).
- Enables the CS 1000 systems to provide an industry-leading PBX feature set on an IP PBX that can be distributed throughout a customers IP network .
- Consolidates voice and data traffic on a single Quality of Service (QoS)-managed network. Network-wide feature transparency is provided using the Nortel Meridian Customer Defined Network (MCDN) protocol.
- Enables Call Servers to work together in a network, over IP facilities, without using circuit switching.

  IP Peer Networking uses direct IP media paths for connections that involve two IP devices. Media streams route directly between the IP Phones and Gateways over the IP network, using Virtual Trunks. This minimizes voice quality issues caused by delay and transcoding between circuit-switched voice and IP packets. For more information on Virtual Trunks, see "Virtual Trunk" (page 21).

SIP and the modified IP Peer Networking feature achieves a direct SIP interface used to interwork with other SIP-enabled Nortel products, such as Multimedia Communication Server 5100 (MCS 5100) and Communication Server 2000 (CS 2000).

SIP is a protocol standard used for establishing, modifying, and terminating conference and telephony sessions in IP networks. A session can be a simple two-way telephone call or it can be a collaborative multimedia conference session. SIP initiates real-time, multimedia sessions which can integrate voice, data, and video. The protocol's text-based architecture speeds access to new services with greater flexibility and more scalability.

IP Peer overlap signaling using the H.323 protocol is also supported.

Nortel does not support the use of a modem in IP networks.

Assumptions

An existing system must be upgraded with CS 1000 Release 5.0 software for IP Peer Networking, and a Signaling Server must be installed and configured to provide SIP or H.323 signaling for Virtual Trunks. SIP and H.323 on the same Signaling Server platform is supported.

The Signaling Server can be any of the following: ISP1100 server (1 GB RAM minimum), CPPM-SS server, IBM or HP Commercial-Off-the-Shelf (COTS) server. The Signaling Server is an industry-standard PC-based server that provides a central processor to drive SIP and H.323 signaling for IP Phones and IP Peer Networking. For more information on the Signaling Server, refer to "Signaling Server" (page 23) and *Signaling Server Installation and Commissioning* (NN43001-312).

To use the Network Routing Service (NRS), a Succession 3.0 H.323 Gatekeeper database must be converted to a CS 1000 Release 4.0 (or later) NRS database. The NRS interface is provided when the Signaling Server is upgraded to CS 1000 Release 4.0 (or later) software. For more information on the NRS, see "Network Routing Service" (page 29) and *Network Routing Service Installation and Commissioning (NN43001-564)* .

A brief overview of the migration procedures is described in *Network Routing Service Installation and Commissioning (NN43001-564)* . However, refer to the Upgrade Procedures NTP appropriate to your system for detailed migration procedures.

> ***Note:*** With the introduction of the NRS, the old Gatekeeper CLI commands are no longer available.

Virtual Trunk

Virtual Trunks are software components configured on virtual loops, similar to IP Phones. A Virtual Trunk acts as the bridge between existing call processing features and the IP network. It enables access to all trunk routing and access features that are part of the MCDN networking feature set. Virtual Trunks do not require dedicated Digital Signal Processor (DSP) resources to provide these features. Virtual Trunks include all the features and settings available to ISDN Signaling Link (ISL)-based TIE trunks, and are configured within trunk routes. Voice Gateway Media Card resources are only allocated for Virtual Trunks when it is necessary to transcode between IP and circuit-switched devices.

> ***Note 1:*** Voice Gateway Media Card is a generic term used when referencing both the ITG-P 24-port Card (dual-slot card) and the 32-port Media Card (single-slot) running the IP Line application. For more information about Voice Gateway Media Cards, refer to *IP Line: Description, Installation, and Operation* (NN43100-500).
>
> ***Note 2:*** Data calls are not supported on Virtual Trunks.

Both SIP and H.323 Virtual Trunks are supported. Up to 1800 Virtual Trunks can be configured on a Signaling Server.

Table 1 "Virtual Trunk limits for each Signaling Server" (page 22) lists the maximum number of Virtual Trunks that can be configured on a Signaling Server.

Table 1
Virtual Trunk limits for each Signaling Server

Protocol	Maximum number of Virtual Trunks
H.323	less than or equal to 1200 (see Note 1)
SIP	1800
Combination of both H.323 and SIP	less than or equal to 1800 (see Note 2)

Note 1: At least 1024 MB of memory is required on the Signaling Server to obtain 1200 H.323 Virtual Trunks.

Note 2: See Table 3 "Maximum number of Virtual Trunk on each Signaling Server" (page 26).

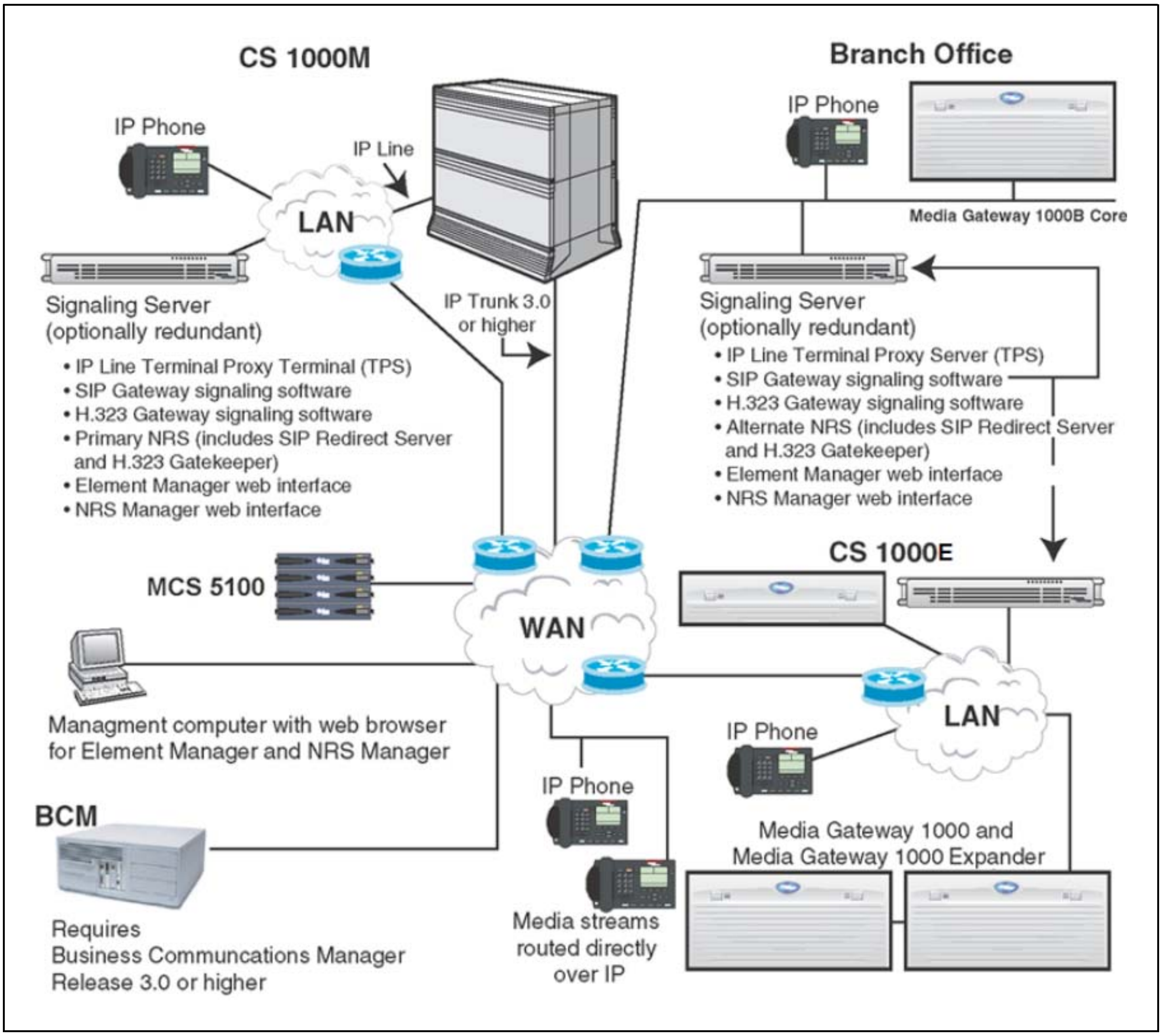
SIP and H.323 Virtual Trunks can reside on the same Signaling Server platform. This is achieved by configuring the Virtual Trunks on separate routes; however, the Virtual Trunks must use the same IP D-channel ID. Each SIP Trunk Gateway occupies one Virtual Trunk route.

Use the Signaling Server Resource Capacity (SSRC) prompt in LD 17 to configure the number of Virtual Trunks on a Signaling Server.

For more information, refer to "Scalability" (page 25) and the Planning and Engineering NTPs.

Figure 1 "An example of IP Peer Networking" (page 23) illustrates an example of an IP Peer Networking configuration.

Figure 1
An example of IP Peer Networking


CS 1000M
IP Phone
IP Line
LAN
Signaling Server
(optionally redundant)
• IP Line Terminal Proxy Terminal (TPS)
• SIP Gateway signaling software
• H.323 Gateway signaling software
• Primary NRS (includes SIP Redirect Server and H.323 Gatekeeper)
• Element Manager web interface
• NRS Manager web interface
IP Trunk 3.0 or higher
Branch Office
IP Phone
Media Gateway 1000B Core
Signaling Server
(optionally redundant)
• IP Line Terminal Proxy Server (TPS)
• SIP Gateway signaling software
• H.323 Gateway signaling software
• Alternate NRS (includes SIP Redirect Server and H.323 Gatekeeper)
• Element Manager web interface
• NRS Manager web interface
MCS 5100
WAN
CS 1000E
Managment computer with web browser for Element Manager and NRS Manager
IP Phone
LAN
BCM
IP Phone
Media Gateway 1000 and Media Gateway 1000 Expander
Requires Business Communcations Manager Release 3.0 or higher
Media streams routed directly over IP

Signaling Server

IP Peer Networking uses a Signaling Server. The Signaling Server can be any of the following: ISP1100 server (1 GB RAM minimum), CPPM-SS server, IBM or HP Commercial-Off-the-Shelf (COTS) server. The Signaling Server provides a central processor to drive SIP and H.323 signaling for IP Phones and IP Peer Networking. The Signaling Server is an industry-standard PC-based server that provides signaling interfaces to the IP network, using software components that operate on the VxWorksTM real-time operating system.

At least one Signaling Server is required for each CS 1000 system. Additional Signaling Servers can be installed in a load-sharing redundant configuration for higher scalability and reliability.

> ***Note:*** The load-sharing redundancy applies only to IP Phones and not to Virtual Trunks.

For more information, refer to *Signaling Server Installation and Commissioning* (NN43001-312).

Applications running on the Signaling Server

The following software components can operate on the Signaling Server:

- IP Line application (UNIStim), including the Line Terminal Proxy Server (LTPS)
- IP Phone Application Server which includes Personal Directory, Callers List, Redial List, and Password administration.

  > ***Note:*** For detailed information on the IP Line application and the IP Phone Application Server, refer to *IP Line: Description, Installation, and Operation* (NN43001-500).

- SIP Gateway signaling software, including IP Peer access and SIP Converged Desktop Service
- H.323 Gateway signaling software for IP Peer access
- Network Routing Service (NRS) comprised of the following components:
  - — SIP Redirect/Registrar Server
  - — H.323 Gatekeeper
  - — Network Connection Service (NCS)
- CS 1000 Element Manager and NRS Manager

  > ***Note 1:*** All the software elements can coexist on one Signaling Server or reside individually on separate Signaling Servers, depending on traffic and redundancy requirements for each element. For details, refer to the *Planning and Engineering* NTPs.
  >
  > ***Note 2:*** If the Signaling Server is running applications other than H.323 or SIP Virtual Trunks, then the maximum number of Virtual Trunks is reduced. If all possible applications are running on the Signaling Server, the maximum number is 382 Virtual Trunks.
  >
  > ***Note 3:*** Refer to the *Planning and Engineering* NTPs for details on the applications that can co-reside on the Signaling Server. There can be limitations to the number of applications that can reside on the Signaling Server at the same time.

The software components are described in the sections that follow.

Scalability

Table 2 "Signaling Server limits" (page 25) shows the capacity limits for each Signaling Server in the network.

Table 2
Signaling Server limits

Signaling Server component	Limit
Network Routing Service (NRS)	100 000 calls per hour 20 000 dialing plan entries 5000 H.323 and/or SIP endpoints
Virtual Trunks	1800 See Table 3 "Maximum number of Virtual Trunk on each Signaling Server" (page 26).

> ***Note:*** Performance degradation occurs if the number of endpoints supported by the NRS exceeds 5000. Degradation, in this case, refers to the increased time required to complete actions such as the following:
>
> - Synchronization between the Primary NRS and the Alternate NRS, and synchronization between the Active NRS and the Failsafe NRS
> - Database actions (such as Commit, Rollback, Automatic Backup, and Restore)
> - Boot-up
>
> However, the ability of the H.323 Gatekeeper to resolve Admission Requests (ARQ) is not affected by an increased number of endpoints.

Both SIP and H.323 Virtual Trunks are supported.

For detailed information on scalability and capacity engineering, refer to the Planning and Engineering NTPs.

- *Communication Server 1000M and Meridian 1: Small System Planning and Engineering* (NN43011-220)
- *Communication Server 1000M and Meridian 1: Large System Planning and Engineering* (NN43021-220)
- *Communication Server 1000E: Planning and Engineering (NN43041-220)* (NN43041-220)

Maximum number of SIP and H.323 Virtual Trunks

The maximum number of SIP and H.323 channels available on each Signaling Server depends on the number of available File Descriptors (FD) for Virtual Trunks. The maximum number of FDs for Virtual Trunks is 1800.

- Each SIP call uses one File Descriptor.
- Each incoming H.323 call uses two File Descriptors.
- Each outgoing H.323 call uses one File Descriptor.

When no more File Descriptors are available (available FD = 0), new channels added on the Call Server will not be able to register on the Signaling Server. Each Signaling Server supports up to 1800 Virtual Trunks. The maximum number of SIP and H.323 trunks depends on traffic patterns, both the split between SIP and H.323 calls and the split between incoming and outgoing H.323 calls. Table 3 "Maximum number of Virtual Trunk on each Signaling Server" (page 26) gives examples of the maximum number of Virtual Trunks supported for different configurations.

Table 3
Maximum number of Virtual Trunk on each Signaling Server

SIP	H.323 (see Note)			Total Virtual Trunks
	Incoming	Outgoing	Total H.323	
1800	0	0	0	1800
0	600	600	1200	1200
0	900	0	900	900
600	0	1200	1200	1800
600	300	600	900	1500

Note: Assumes H.245 tunneling is enabled.

The formula to calculate the maximum number of Virtual Trunks is:

> (Num_of_SIP x 1 FD) + (Num_of_Incoming_H323 x 2 FD) +
> (Num_of_Outgoing_H323 x 1 FD) <= Max_Num_of_FDs
> where Max_Num_of_FDs = 1800

Impact of H.245 tunneling By default, H.245 tunneling is enabled. Unless there is a specific reason to disable tunneling, such as for maintenance, it should always be enabled. When tunneling is disabled, the handling capacity of the Signaling Server is reduced to a maximum of 900 H.323 Virtual Trunks. See "H.245 tunneling" (page 38).

Terminal Proxy Server

The Terminal Proxy Server (TPS) is a SIP/ H.323 signaling proxy software component for IP Phones. The TPS supports up to 5000 IP Phones on each Signaling Server. The TPS, in conjunction with the Call Server, delivers a full suite of telephone features.

IP Peer Networking supports the following telephones for IP telephony:

- Nortel IP Phone 1110
- Nortel IP Phone 1120E
- Nortel IP Phone 1140E
- Nortel IP Phone 1150
- Nortel IP Phone 2001
- Nortel IP Phone 2002
- Nortel IP Phone 2004
- Nortel IP Phone 2007
- Nortel IP Audio Conference Phone 2033
- Nortel IP Softphone 2050
- Nortel Mobile Voice Client 2050
- Nortel WLAN Handset 2210
- Nortel WLAN Handset 2211
- Nortel WLAN Handset 2212

You can configure each IP Phone to use the Dynamic Host Configuration Protocol (DHCP) to register with a Call Server for feature control.

SIP Gateway Signaling software

The SIP Gateway offers an industry-standard SIP-based IP Peer solution. SIP Gateway delivers a SIP interface for interoperability with Nortel SIP products and other industry SIP-based products.

SIP Gateway is a generic term used to refer to the SIP IP Peer networking application. The SIP Trunk Gateway provides a direct trunking interface between the CS 1000 systems and a SIP domain. The SIP Trunk Gateway application resides on a Signaling Server and has two functions:

- acts as a SIP User Agent, which services one or more end users in making/receiving SIP calls
- acts as a signaling gateway for all CS 1000 telephones (IP Phones, analog [500/2500-type] telephones, and digital telephones), which maps ISDN messages to and from SIP messages

As the call-signaling gateway, the SIP trunking application does the following:

- maps telephony numbers to and from SIP Uniform Resource Identifiers (URIs)
- performs client registration
- maps ISDN messages to and from SIP messages
- establishes the speech path between the desktop and SIP endpoints

> ***Note:*** SIP endpoints are also known as SIP User Agents that service one or more endpoints. This document uses the term "SIP endpoints".

The SIP Trunk Gateway is implemented according to SIP standards. The SIP Trunk Gateway can connect two CS 1000 nodes and can also connect CS 1000 systems to other Nortel or third-party SIP-enabled products. This direct SIP interface is used to interwork with products such as MCS 5100.

The direct SIP interface provides the following:

- removes the requirement for a SIP/PRI gateway between the CS 1000 and the MCS 5100 systems
- improves voice quality through peer-to-peer communication of IP devices

SIP connectivity (also known as SIP trunking) provides a direct media path (trunk interface) between a user in the CS 1000 domain and a user in a SIP domain.

SIP Converged Desktop Service

The SIP Converged Desktop Service (SIP CDS) is a CS 1000 Release 4.0 (or later) and MCS 5100 Release 3.0 (or later) feature. SIP CDS brings multimedia features to CS 1000 users. SIP CDS allows a user to have access to multimedia features on MCS 5100 and voice features on CS 1000 systems at the same time. SIP CDS allows users to use their existing telephony system for voice communication and to use their PC for multimedia communication.

H.323 Gateway Signaling software

H.323 Gateway Signaling software provides the industry-standard H.323 protocol, to provide connectivity to H.323 Gateways and circuit switches that act as H.323 Gateways. H.323 Gateway Signaling software uses an H.323 Gatekeeper to resolve addressing for systems at different sites. The H.323 Gateway uses Virtual Trunks to enable direct, end-to-end voice paths between two IP devices.

Direct IP media paths provide the following benefits:

- elimination of multiple IP Telephony to circuit-switched conversions

- improved voice quality
- simplified troubleshooting

See "Interworking protocols" (page 32) for further information.

Overlap Signaling

Overlap signaling over IP is supported using the H.323 protocol.

> ***Note:*** Overlap signaling is not supported using the Session Initiation Protocol (SIP).

In the H.323 network, dialed digits can be sent out or received in either en bloc (normal dialing) or overlap modes. Overlap signaling is sending some digits of the called-party number in the first signaling message (SETUP messages) followed by further digits in subsequent signaling messages (INFORMATION messages). Overlap signaling improves call setup time.

For detailed information, refer to "Overlap signaling" (page 251).

Network Routing Service

IP Peer Networking uses the NRS to simplify the configuration of IP component addressing. The NRS (which is optionally redundant) manages a centralized numbering plan for the network. The NRS allows customers to manage a single network dialing plan for SIP, H.323, and mixed SIP/H.323 networks.

> ***Note:*** Within each Call Server, configure the numbering plan information required for the Call Server software to internally route calls, such as routing information for locally accessible numbers.

The IP Peer Networking feature provides the NRS where all CS 1000 systems in the network can register. This eliminates the need for manual configuration of IP addresses and numbering plan information at every site.

The VxWorks-based NRS combines the following:

- SIP Redirect Server (see "SIP Redirect Server software" (page 30)) and SIP Registrar (see "SIP Registrar" (page 30))
- H.323 Gatekeeper (see "H.323 Gatekeeper software" (page 31))
- Network Connection Service (NCS) (see "H.323 Gatekeeper software" (page 31))

The SIP Redirect Server and H.323 Gatekeeper can reside on the same Signaling Server. The data entry for the dialing plan is common for both SIP and H.323. The Network Routing Service (NRS) Manager includes both the SIP Redirect Server and the H.323 Gatekeeper.

The NRS can operate in two modes:

- Stand-alone mode — The host Signaling Server does not have an attached Call Server. During installation of a stand-alone Signaling Server, the Call Server IP address defaults to 0.0.0.0.
- Co-resident mode — The host Signaling Server has an attached Call Server. The Signaling Server is running the NRS as well as other applications such as the IP Line TPS and Gateway Signaling Software. Refer to "Applications running on the Signaling Server" (page 24).

The Alternate NRS is supported only on a Leader Signaling Server. Nortel recommends that, for network reliability, the Alternate NRS be located in a physical location separate from the Primary NRS.

For more information, see *Network Routing Service Installation and Commissioning (NN43001-564)* .

SIP Redirect Server software

Building on the H.323 Gatekeeper, the SIP Redirect Server is used to facilitate centralized dialing plan management and the configuration of the network routing information for the SIP domain.

Nortel has many products with a SIP interface. A SIP Redirect Server translates telephone numbers recognized by Enterprise Business Network (EBN) voice systems to IP addresses in the SIP domain. As a result, the SIP Redirect Server interfaces with SIP-based products.

The SIP Redirect Server resides on the Signaling Server. The SIP Redirect Server is used to interconnect with other Nortel communication servers using SIP. Along with the H.323 Gatekeeper application, the SIP Redirect Server has access to the endpoint/location database. The SIP Redirect Server has the ability to access the CS 1000 system’s location database in order to direct SIP Trunk Gateways and SIP Phones within the networked environment.

SIP Registrar

The SIP Registration Server is also known as the SIP Registrar. Registration is one way that the server can learn the location of a user (SIP client). The SIP Registrar accepts registration requests from SIP Phones, SIP Trunk Gateways, and other certified compatible third-party SIP user agents that are supported.

Upon initialization, and at periodic intervals, a user’s telephone sends REGISTER messages to the SIP Registrar in the same domain. The contact information from the REGISTER request is then made available to other SIP servers, such as proxies and redirect servers, within the same administrative domain. The registration process precedes the call setup.

The SIP Registrar is collocated with the SIP Redirect Server on the Signaling Server.

By storing information mapping device addresses on a SIP Registrar, communication can be addressed to a person's name instead of a complex number scheme. A person simply registers one or more SIP devices (for example, a SIP Phone) with the network and becomes reachable, wherever he or she may be, independent of the details of the networks and devices involved.

For more information, refer to *Network Routing Service Installation and Commissioning (NN43001-564)* .

H.323 Gatekeeper software

The H.323 Gatekeeper manages a centralized numbering plan for the H.323 network. This enables simplified management of the CS 1000 network. The H.323 Gatekeeper software identifies the IP addresses of H.323 Gateways, based on the network-wide numbering plan, in the CS 1000 systems and third-party systems.

Network Connection Server

The NRS also includes the Network Connection Service (NCS). The NCS is used for the Branch Office (including the Survivable Remote Gateway [SRG]), IP Line Virtual Office, and Geographic Redundancy features. The NCS allows the Line TPS (LTPS) to query the NRS using the UNIStim protocol. For more information, see *Network Routing Service Installation and Commissioning (NN43001-564)* .

Element Manager web interface

Element Manager is a simple and user-friendly web-based interface that supports a broad range of system management tasks, including:

- configuration and maintenance of IP Peer and IP telephony features
- configuration and maintenance of traditional routes and trunks
- configuration and maintenance of numbering plans
- configuration of Call Server data blocks (such as configuration data, customer data, Common Equipment data, D-channels) maintenance commands, system status inquiries, backup and restore functions
- software download, patch download, patch activation

Element Manager has many features to help administrators manage systems with greater efficiency. For example:

- Web pages provide a single point-of-access to parameters that were traditionally available through multiple overlays.

- Parameters are presented in logical groups to increase ease-of-use and speed-of-access.
- The "hide or show information" option enables administrators to see information that relates directly to the task at hand.
- Full-text descriptions of parameters and acronyms help administrators reduce configuration errors.
- Configuration screens offer pre-selected defaults, drop-down lists, check boxes, and range values to simplify response selection.

The Element Manager web server resides on the Signaling Server and can be accessed directly through a web browser or Telephony Manager (TM). The TM navigator includes integrated links to each network system and their respective instances of Element Manager.

NRS Manager web interface

The NRS Manager is the web interface for the NRS. The web interface is common to both the H.323 Gatekeeper and the SIP Redirect Server. NRS Manager is used for populating the location and registration database. For detailed information, refer to *Network Routing Service Installation and Commissioning (NN43001-564)* .

Interworking protocols

Peer-to-peer call and connection control at the IP level requires peer-to-peer protocol. IP Peer Networking uses the SIP and H.323 protocols.

To support traditional PBX signaling on an IP network, it can be necessary to transport non-IP peer signaling information from peer to peer. This is achieved by "tunneling" the legacy protocol in the IP peer protocol.

SIP, H.323, and MCDN tunneling is supported.

Session Initiation Protocol

Session Initiation Protocol (SIP) is supported by CS 1000, which complies with the standards described in the following Request for Comments (RFC) Internet Engineering Task Force (IETF) standards documents:

- RFC 3261 – SIP: Session Initiation Protocol
- RFC 3262 – Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
- RFC 2806 – URLs for Telephone Calls
- RFC 3264 – An Offer/Answer Model with the Session Description Protocol (SDP)
- RFC 3265 – Session Initiation Protocol (SIP)-Specific Event Notification

- RFC 3311 – The Session Initiation Protocol (SIP) UPDATE Method
- RFC 2976 – The SIP INFO Method

SIP is an Application Layer (Layer 7 of the OSI Reference Model) protocol used for establishing, modifying, and terminating real-time conference and telephony sessions over IP-based networks. SIP uses text-based messages, much like Hypertext Transfer Protocol (HTTP) and Simple Mail Transfer Protocol (SMTP). SIP also uses Session Description Protocol (SDP) for media description.

A SIP session is any interactive communication that takes place between two or more entities over the IP network, from a simple two-way telephone call or instant message to a collaborative multimedia conference session.

SIP is a simple, transport-independent, text-based protocol used for multimedia call control and enhanced telephony services. SIP has only six different method types. These methods, when combined, allow for complete control over a multimedia call session while limiting complexity. SIP is transport-layer independent. Both TCP and UDP can be used as the transport protocol for SIP; however, TCP is the default mechanism.

> ***Note:*** Nortel recommends that customers use TCP as the transport protocol for SIP traffic.

SIP is text-based in that a method is formed using a textual header with fields that contain call properties. This text-based approach is easy to parse, has small packet overhead, and is flexible.

SIP clients are also known as SIP User Agents. These clients communicate with SIP servers in a client-server fashion. User Agents also act as servers when the SIP request reaches its final destination. These user agents contain the full SIP state machine and can be used without intermediate servers.

Table 4 "SIP components" (page 33) lists and describes the SIP components.

Table 4
SIP components

Component	Description
SIP User Agent	The end system component for the call
SIP Network Server	The network device that handles the signaling associated with multiple calls

SIP User Agent

The User Agent has a client and server element.

- User Agent Client — the client element initiates the calls
- User Agent Server — the server element answers the calls

Peer-to-peer calls can, therefore, be made using a client-server protocol.

SIP Network Server

The main function of the SIP Network Server is to provide name resolution and user location, as the caller is unlikely to know the IP address or host name of the called party.

An *e-mail-like* address or a telephone number is usually associated with the called party. Using this information, the caller’s User Agent identifies with a specific server to resolve the address information.

Three forms of SIP Network Server can exist in a network: the SIP stateful proxy server, the SIP stateless proxy server, and the SIP redirect server. The three forms function as follows:

- A SIP proxy server (both stateful and stateless) receives requests, determines where to send the requests, and passes them on to the next server.
  - stateful proxy — a proxy server in a stateful mode remembers the incoming requests it receives, along with the responses it sends back and the outgoing requests it sends on
  - stateless proxy — a proxy server acting in a stateless mode forgets all information once it has sent a request
- A SIP redirect server receives requests, but does not pass the requests onto the next server. Instead, the SIP redirect server sends a response back to the caller, indicating the address for the called user. Because the response includes the address of the called user, the caller can then directly contact the called party at the next server.

CS 1000 Release 5.0 NRS is offered in two versions: (1) a Vxworks-based NRS comprising a SIP Redirect Server, Network Connect Server (NCS), and H.323 Gatekeeper (GK) ; (2) a Linux-based NRS comprising a SIP Proxy Server and Redirect Server, NCS, and GK. The VxWorks-based NRS application software can be run on any of the following server platforms: ISP1100 server (1 GB RAM minimum), CP-PM server, IBM or HP Commercial-Off-the-Shelf (COTS) server. On any of those four server platforms the VxWorks-based NRS can be configured to run either stand-alone or co-resident with other CS 1000 Signalling Server applications, i.e. UNIStim IP Phone Line Terminal Proxy (LTPS), IP Peer virtual trunk SIP or H.323 signaling Gateway (VTRK GW), IP Phone

Application Server. There are no functional changes in the Vxworks-based NRS application in CS 1000 Rls 5.0, as compared with CS 1000 Rls 4.5. The Linux-based NRS application software can be run on either IBM or HP COTS server. The CS 1000 Rls 5.0 Signalling Server applications cannot be run on the Nortel Linux-based COTS server, therefore the Linux-based NRS can be configured to run only stand-alone.

SIP addressing is built around either a telephone or a web host name. For example, the SIP address can be based on a URL such as the following: SIP:john.doe@companyabc.com. The format makes it very easy to guess a SIP URL based on an e-mail address. The URL is translated into an IP address through a Domain Name Server (DNS).

SIP negotiates the features and capabilities of the session at the time the session is established. With SIP, a common set of audio and video compression algorithms negotiate prior to establishing the SIP session. This advance negotiation reduces the call setup time (compared to the time required for H.323 sessions). The Session Description Protocol (SDP) is used for this advance negotiation process. Once the session is established, the designated capabilities can be modified during the call. For example, additional features can be added if both terminals are capable and can negotiate a common compression algorithm.

SIP supports both unicast (one-to-one) sessions and multicast (one-to-many) communication.

SIP/MCDN

SIP services also implement tunneling of MCDN messages. Tunneling enables preservation of MCDN features if calls between two CS 1000 systems are over a SIP trunk or the call is redirected back to the CS 1000 systems from MCS 5100.

If MCS 5100 tunnels MCDN messages, Trunk Route Optimization (TRO) removes the unnecessarily used DSP/Virtual Trunk channels between CS 1000 and MCS 5100 systems. The result is a significant cost reduction and voice quality improvement for the converged desktop users.

MCDN tunneling is supported over SIP Virtual Trunks. However, if calls are connected between two CS 1000 systems using the MCS 5100, then the SIP trunk between two CS 1000 systems does not support the full set of MCDN features unless the proxy that connects the two systems can tunnel the MCDN messages.

> ***Note 1:*** While the MCDN protocol is supported by MCDN tunneling in SIP, QSIG is not supported by CS 1000 in terms of Q.SIG over SIP.

> ***Note 2:*** SIP uses a subset of the MCDN content in UIPE format and carries it like H.323 does; however, this is only for information that does not have standardized transport mechanisms.

For detailed information about SIP, refer to RFC 3261.

H.323 protocol

CS 1000 systems support H.323 version 4.0.

H.323 is the leading standard in the Voice over IP (VoIP) area. The term VoIP stands for more than only voice transmission in IP networks. It covers an abundance of applications that are now being successively integrated due to the universality and ubiquity of the IP networks. Enhanced performance of IP and Ethernet networks, as well as the improved manageability of the bandwidth, allow traditional switched-network applications — such as Automatic Call Distribution, Real-time Messaging and Teleworking — to be offered in IP networks.

In addition to voice applications, H.323 provides mechanisms for video communication and data collaboration, in combination with the International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) T.120 series of standards. The H.323 standard (published in 1996 by the ITU-T) represents the basis for data, voice, and video communication over IP-based LANs and the Internet.

The H.323 standard refers to many other standards such as H.245, H.225, H.450. H.323 regulates the technical requirements for visual telephony, which means the transmission of audio and video in packet-based networks. Because IP is the prevailing protocol in packet-based networks (with about 90 percent market share), the H.323 standard is interpreted as a standard for multimedia communication in IP networks.

By definition, H.323 focuses on IP packet-based networks that do not provide any guaranteed service quality. For example, packets can be lost and real-time (voice and video) traffic does not take precedence over non-real-time, and therefore delay-insensitive, data traffic.

Recent developments in IP networking technology introduce Quality of Service (QoS) mechanisms that lead to improved voice/video quality. However, because the majority of IP networks today still do not have QoS capabilities, the H.323 mechanisms help provide reliable communication.

Because IP runs on any existing Layer 2 technologies, H.323 can be used over:

- Ethernet
- Fast Ethernet
- Gigabit Ethernet

- FDDI
- Token-Ring

Recent implementation proves that H.323 can also be used beyond LANs, in multi-site configurations over Wide Area Networks (WANs) based on T1, Frame Relay, and ATM technology.

H.323 is often characterized as an "umbrella specification" because it refers to various other ITU standards. The topology and its parts, as well as the protocols and standards, are specified in H.323.

Table 5 "H.323 components" (page 37) lists and describes the H.323 components.

Table 5
H.323 components

Component	Description
Terminal	Terminals represent the end devices of every connection.
Gateway	Gateways establish the connection in other networks. That is, gateways connect the H.323 network with the switched network of PBXs and Central Office switches.
Gatekeeper	Gatekeepers take over the task of translating between telephone numbers (for example, in accordance to the E.164 numbering standard) and IP addresses. Gatekeepers also manage the bandwidth and provide mechanisms for terminal registration and authentication.
Multipoint Control Units (MCUs)	MCUs are responsible for establishing multipoint conferences. The H.323 standard makes the distinction between callable and addressable end devices: all components are addressable; gatekeepers are, however, not callable.

The four components communicate by exchanging information flows among each other. The information flows are split into five categories:

- Audio (digitized and coded) voice
- Video (digitized and coded full-motion image communication)
- Data (files such as text documents or images)
- Communication control (such as exchange of supported functions and controlling logical channels)
- Controlling connections (such as connection setup and connection release)

H.323/MCDN

MCDN tunneling in H.323 is supported.

Internet-enabled Meridian 1 Systems also support MCDN tunneling in H.323, using IP Trunk 3.0 (or later), which supports H.323 Gatekeeper operation, as well as non-call associated signaling.

Wireless LAN interworking (802.11 Wireless IP Handsets)

802.11 Wireless IP Handsets use H.323 as a protocol to access a Call Server, as opposed to using H.323 to access an H.323 network. For the 802.11 Wireless IP Handset, the H.323 network consists of the 802.11 Wireless IP Gateway to which it terminates, instead of the entire H.323 network. The Call Server sees 802.11 Wireless IP Handsets as ordinary telephones.

802.11 Wireless IP Handsets can access Virtual Trunk routes like any other terminal device; however, indirect media paths are used. Also, no direct media connection occurs between 802.11 Wireless IP Handsets and IP Phones or Media Gateways (the media stream from the 802.11 Wireless IP Handset terminates at its IP Line/Trunk).

Call independent signaling connection and connectionless transport

With IP Peer Networking, signals cannot be sent directly from endpoint to endpoint without first determining the signaling IP address of the remote endpoint, using standard Gatekeeper procedures. This requires setting up an end-to-end path or connection to support the messaging. However, the base MCDN Peer-to-Peer signaling, used to provide supplementary service signaling independently of any established calls, uses connectionless signaling; it does not use a path.

Therefore, connectionless MCDN Non-Call Associated Signaling (NCAS) is transported as though it is a virtual, path-oriented connection (virtual call) using the H.323 call-independent call-signaling connection. Because this call is essentially an H.323 call with no media, standard H.323 Gatekeeper procedures apply. As a direct result, MCDN services using connectionless transport between the Call Server and the Signaling Server are not transported over the IP network using H.323 connectionless transport.

Alternate routing is not supported for NCAS messages over IP Peer. Services such as Network Ring Again, Network ACD and Centralized CallPilot that rely on NCAS may not work over alternate routes if the primary IP Peer route fails.

H.245 tunneling

H.245 tunneling is supported, and is enabled by default. This conserves resources, synchronizes call signaling and control, and reduces call setup time. If required, the user has the option to turn the tunneling on and off. This is done using CLI commands through the VxWorks shell on the Signaling Server.

H.245 specifies the signaling protocol which is used to:

- establish a call
- determine the capabilities of a call
- issue the commands necessary to open and close media channels

The H.245 control channel is responsible for control messages governing the operations of H.323 terminals.

H.245 tunneling enables the reuse of socket FDs used for H.323 call signaling. The H.245 control messages are sent on the same TCP link that was opened for the H.225 call control message exchange with the peer node. This halves the number of sockets used for each call.

Number of supported Virtual Trunks with H.245 tunneling enabled

If H.245 tunneling is enabled (the default), then the following are supported on the Signaling Server:

- up to 1200 H.323 Virtual Trunks
- up to 1800 SIP Virtual Trunks
- a combination of both H.323 and SIP Virtual Trunks

  If there is a combination of H.323 and SIP trunks, then the available number of Virtual Trunks is shown in the following calculation:

  1800 - [(1 x H.323 channels) + SIP channels]
  (where 1800 is the maximum number of Virtual Trunks)

  Example 1: 1200 H.323 and 600 SIP

  1800 - [(1 x 1200 H.323 channels) + 600 SIP channels]
  = 0 available Virtual Trunks

  Example 2: 900 H.323 and 900 SIP

  1800 - [(1 x 900 H.323 channels) + 900 SIP channels]
  = 0 available Virtual Trunks

  Example 3: 0 H.323 and 1800 SIP

  1800 - [(1 x 0 H.323 channels) + 1800 SIP channels]
  = 0 available Virtual Trunks

  Example 4: 1200 H.323 and 0 SIP

  1800 - [(1 x 1200 H.323 channels) + 0 SIP channels]
  = 600 available Virtual Trunks

  > ***Note:*** The 600 available trunks must be SIP trunks, as the number of H.323 channels is already at the maximum limit of 1200.

Number of supported Virtual Trunks with H.245 tunneling disabled

If H.245 tunneling is disabled, then the following are supported on the Signaling Server:

- up to 900 H.323 Virtual Trunks
- up to 1800 SIP Virtual Trunks
- a combination of H.323 and SIP trunks

  If there is a combination of H.323 and SIP trunks, then the available number of Virtual Trunks is shown in the following calculation:

  1800 - [(2 x H.323 channels) + SIP channels]
  (where 1800 is the maximum number of Virtual Trunks)

  Example 1: 900 H.323 and 0 SIP

  1800 - [(2 x 900 H.323 channels) + 0 SIP channels]
  1800 - [(1800 H.323 channels) + 0 SIP channels]
  = 0 available Virtual Trunks

  Example 2: 600 H.323 and 600 SIP

  1800 - [(2 x 600 H.323 channels) + 600 SIP channels]
  1800 - [(1200 H.323 channels) + 600 SIP channels]
  = 0 available Virtual Trunks

  Example 3: 0 H.323 and 1800 SIP

  1800 - [(1 x 0 H.323 channels) + 1800 SIP channels]
  = 0 available Virtual Trunks

SIP signaling

Contents

This section contains information on the following topics:

Introduction

The SIP Trunk Gateway offers an industry-standard SIP-based IP Peer solution. A SIP Trunk Gateway delivers a SIP interface for interoperability with Nortel SIP products and other industry SIP-based products.

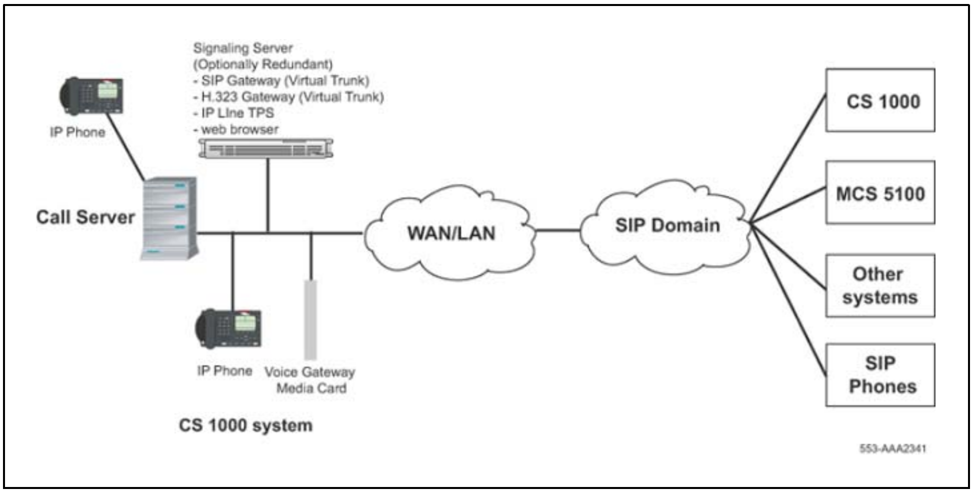
The SIP Trunk Gateway is implemented according to SIP standards. The SIP Trunk Gateway can connect two CS 1000 nodes and can also connect CS 1000 systems to other Nortel or third-party SIP-enabled products. This SIP Trunk Gateway interworks with the MCS 5100 system.

The SIP trunking application resides on the Signaling Server. The SIP Trunk Gateway provides a direct trunking interface between the CS 1000 systems and a SIP domain.

For information, see "SIP Trunk Gateway software — trunk route redundancy" (page 158).

Figure 2 "CS 1000 SIP Trunk Gateway interworking" (page 42) shows the CS 1000 SIP Trunk Gateway interworking.

Figure 2
CS 1000 SIP Trunk Gateway interworking



The direct SIP interface provides the following:

- removes the requirement for a SIP/PRI gateway between the MCS 5100 and the CS 1000 systems
- improves voice quality through peer-to-peer communication of IP devices

SIP connectivity (also known as SIP trunking) provides a direct media path (trunk interface) between a user in the CS 1000 domain and a user residing in a SIP domain.

SIP trunking (the SIP Trunk Gateway) acts as a SIP User Agent and a call-signaling gateway for the telephones (analog [500/2500-type] telephones, digital telephones, and IP Phones).

- As a SIP User Agent, it services one or more end users in making and receiving SIP calls.
- As a call-signaling gateway, the SIP trunking application does the following:
  - maps telephony numbers to and from SIP Uniform Resource Identifiers (URIs)
  - performs client registration
  - maps ISDN messages to and from SIP messages
  - establishes the speech path between the desktop and SIP endpoints

SIP requests and responses

Table 6 "SIP request methods" (page 43) shows the SIP request methods.

Table 6
SIP request methods

Method	Description
INVITE	Indicates a user or service is being invited to participate in a call session. A re-INVITE message is an INVITE message that is used after a call is answered.
ACK	Confirms that the client has received a final response to a request.
BYE	Terminates a call and can be sent by either the caller or the called party.
CANCEL	Cancels any pending searches but does not terminate a call that has already been accepted.
OPTIONS	Queries the capabilities of servers.
REFER	Provides a mechanism allowing the party sending the message to be notified of the outcome of the referenced request. This can be used to enable many applications, including call transfer.
UPDATE	Allows a client to update parameters of a session (such as the set of media streams and their codecs) but has no impact on the state of a dialog. In that sense, it is like a re-INVITE message, but unlike re-INVITE, it can be sent before the initial INVITE has been completed.
INFO	Carries session-related control information during a session.
PRACK	Provides reliable provisional response messages.
SUBSCRIBE /NOTIFY	Requests notification from remote nodes indicating that certain events have occurred.

Table 7 "SIP response methods" (page 43) shows the SIP response methods.

Table 7
SIP response methods

Response numbers	Type of response
SIP 1xx	Informational responses
SIP 2xx	Successful responses
SIP 3xx	Redirection responses
SIP 4xx	Client Failure responses
SIP 5xx	Server Failure responses
SIP 6xx	Global Failure responses

Format of a SIP message

A SIP message consists of the following components:

- start line
- one or more header fields
- an empty line indicating the end of message header
- an optional message body

A start line can be either a request line or a response line:

- A request line distinguishes a request message.
- A response line distinguishes a response message.

Request line

A request line is defined as follows:

Method <space> Request-URI <space> SIP-Version <CRLF>

For example: INVITE sip:john@myServiceProvider.com SIP/2.0

In this example, INVITE is the method, followed by user URI sip:john@myServiceProvider.com, and followed by SIP version.

Response line

A response line is defined as follows:

SIP-Version <space> Status-Code <space> Reason-Phrase <CRLF>

For example: SIP/2.0 100 Trying

In this example, SIP/2.0 is the version string, 100 is the status code, and "Trying" is the text description of status code.

Direct IP Media Paths

With IP Peer Networking, the SIP Trunk Gateway signaling software enables direct IP voice paths to IP devices. An endpoint is the SIP Trunk Gateway that terminates a SIP signaling stream. A SIP Trunk Gateway that terminates SIP signaling registers at the NRS (specifically the SIP Redirect Server in the NRS) as an endpoint. IP Phones interact with the SIP Trunk Gateway software to appear as SIP devices that support Direct IP Media Paths.

> ***Note 1:*** IP Peer Networking supports both Media Gateways and third-party Gateways that have been tested for compatibility. Use the Gateway to enable communication between an H.323 or SIP network and circuit-switched equipment. Interfaces provided by Media Gateways

operate in H.323/SIP standard mode and support MCDN feature capabilities. They operate autonomously in the network.

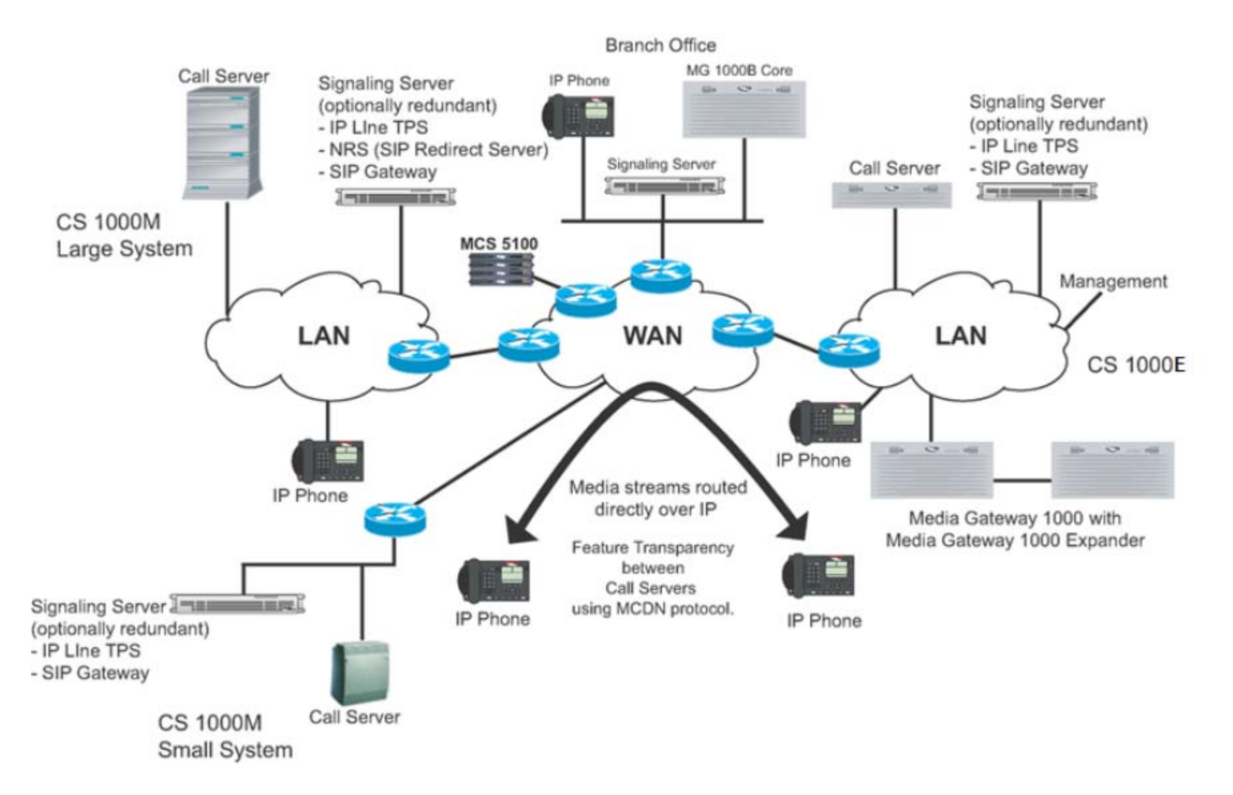
Note 2: A Media Gateway is a gateway that uses a protocol similar to the Media Gateway Control Protocol (MGCP). The Media Gateway houses peripheral cards. Media Gateways are controlled directly by the Call Server. Peripheral cards are housed in the Intelligent Peripheral Equipment (IPE) shelf in CS 1000M Systems.

The Direct IP Media Path functionality ensures that when any IP device in the network (for example, an IP Phone) connects to another IP address (for example, an IP Phone), the media path uses direct IP connections and does not pass through a central circuit-switched PBX. When the connection is made between a Virtual Trunk and a circuit-switched device (for example, a PRI trunk), a Digital Signal Processor (DSP) resource on the Voice Gateway Media Card is allocated to transcode the media stream from IP to circuit-switched.

When the network address of the local IP device or DSP resource is determined, the address is signaled over standard SIP to the far end so a direct media path can be established. If a call-modification operation is involved (for example, Call Transfer), further signaling of the address information occurs using the SIP re-INVITE or UPDATE methods.

Figure 3 "An example of IP Peer Networking using Virtual Trunk and direct media paths" (page 46) shows a media path routed directly over IP, not using a circuit switch.

Figure 3
An example of IP Peer Networking using Virtual Trunk and direct media paths



IP Phone to IP Phone (on separate Call Servers)

An IP Phone at Site A calls an IP Phone at Site B (see Figure 4 "User A dials User B" (page 47)). When the user presses a key on the IP Phone, a signaling message is carried over the IP network.

The Call Server on the originating node selects an ISDN route and a virtual IP trunk, based on the dialed digits translation. After terminating on a Virtual Trunk, D-channel signaling occurs over IP. This includes basic call setup signals (ISDN over IP, as well as Nortel MCDN signaling over IP, which is used for networking features). The ISDN signaling is converted to a SIP message by the SIP Trunk Gateway on the Signaling Server. MCDN messages are carried within the SIP message, using proprietary SIP message-body extensions.

On the terminating node, the SIP signaling is received at the SIP Trunk Gateway on the Signaling Server. The SIP message is converted to an ISDN message which is then sent to the Call Server. The terminating Call Server translates the received digits to an IP Phone DN. When the terminating IP Phone answers the call, the terminating node returns an ISDN CONNECT message, then converts the ISDN message to the SIP 200 OK message. The Signaling Servers complete the exchange of the IP media information required to establish the IP media path. The originating and terminating Call Servers establish a direct two-way IP media path between the two IP Phones.

Basic network call walk-through

When a user makes a call on a CS 1000 system, the dialed digits are translated to determine if the user is attempting to reach an internal or external telephone.

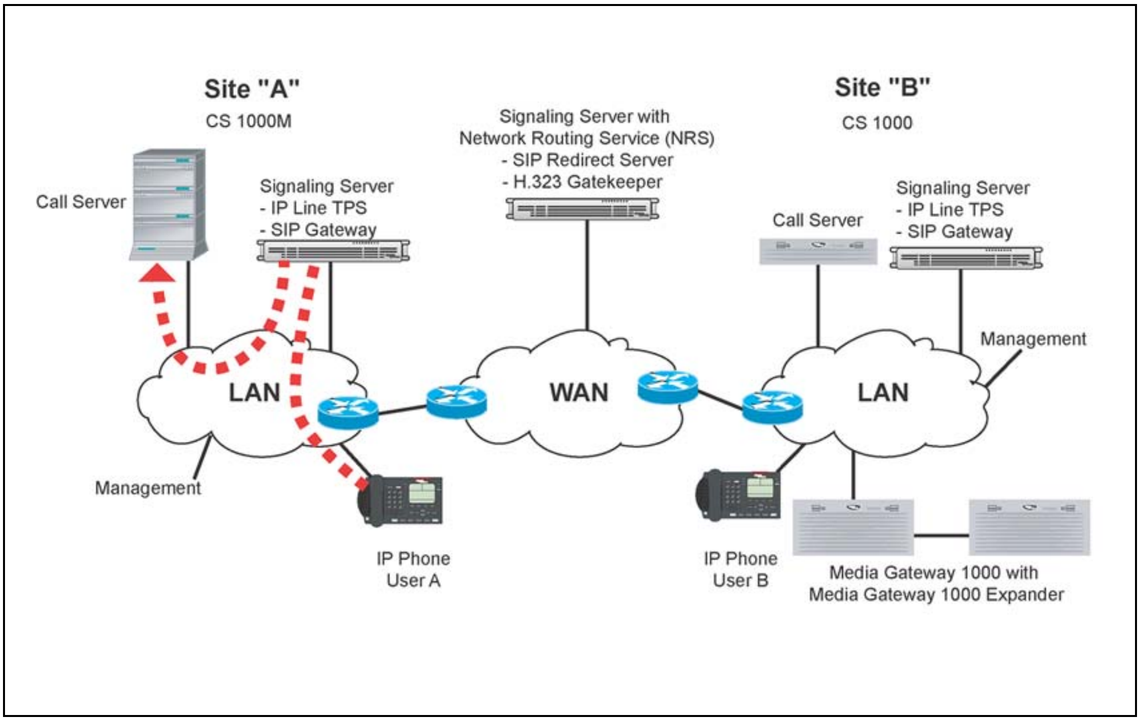
If the user is attempting to reach an internal telephone, the call is terminated on the internal device. When the system determines that the user is attempting to reach a telephone or service using the IP network, the call routes to the SIP Trunk Gateway software. The SIP Trunk Gateway software uses the NRS, specifically the SIP Redirect Server, to help with call routing.

> ***Note:*** Only the primary messages are illustrated in the following call flows.

The following scenario describes the Direct IP Media Path functionality for a basic network call:

1. User A on Call Server A dials the DN of User B on Call Server B. Call Server A collects the digits through the Terminal Proxy Server (TPS) on Signaling Server A. See Figure 4 "User A dials User B" (page 47).

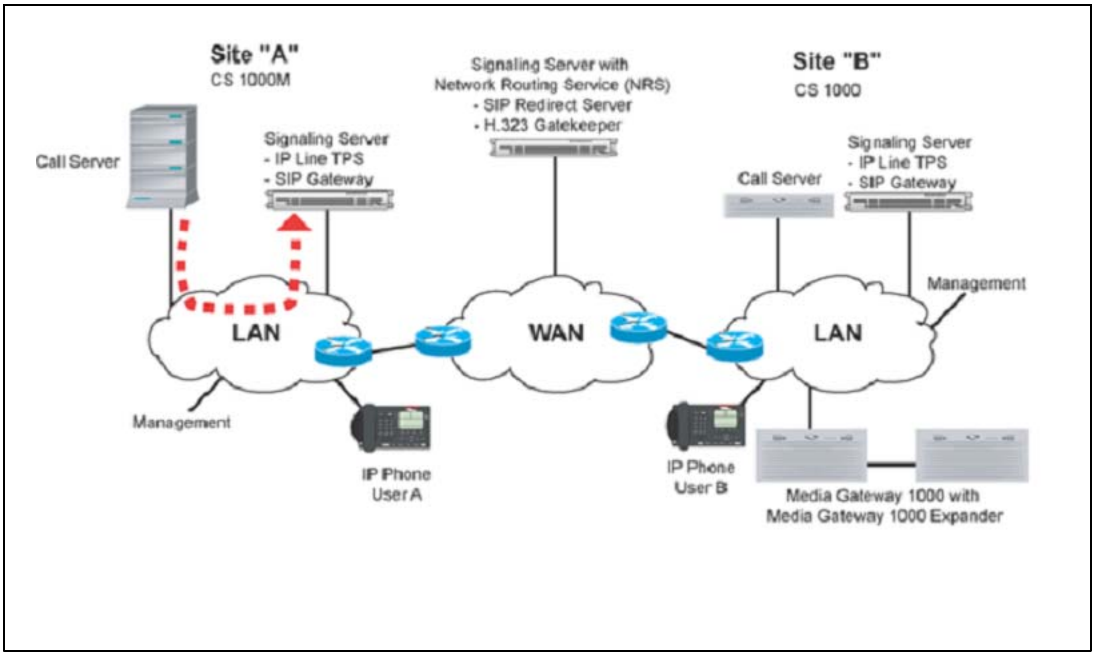
Figure 4
User A dials User B



2. Call Server A determines that the dialed DN is at another site. Call Server A selects the codec list, allocates bandwidth, and routes the call to the SIP Trunk Gateway using the Virtual Trunk. See Figure 5 "Call Server A routes the call to the IP network" (page 48).

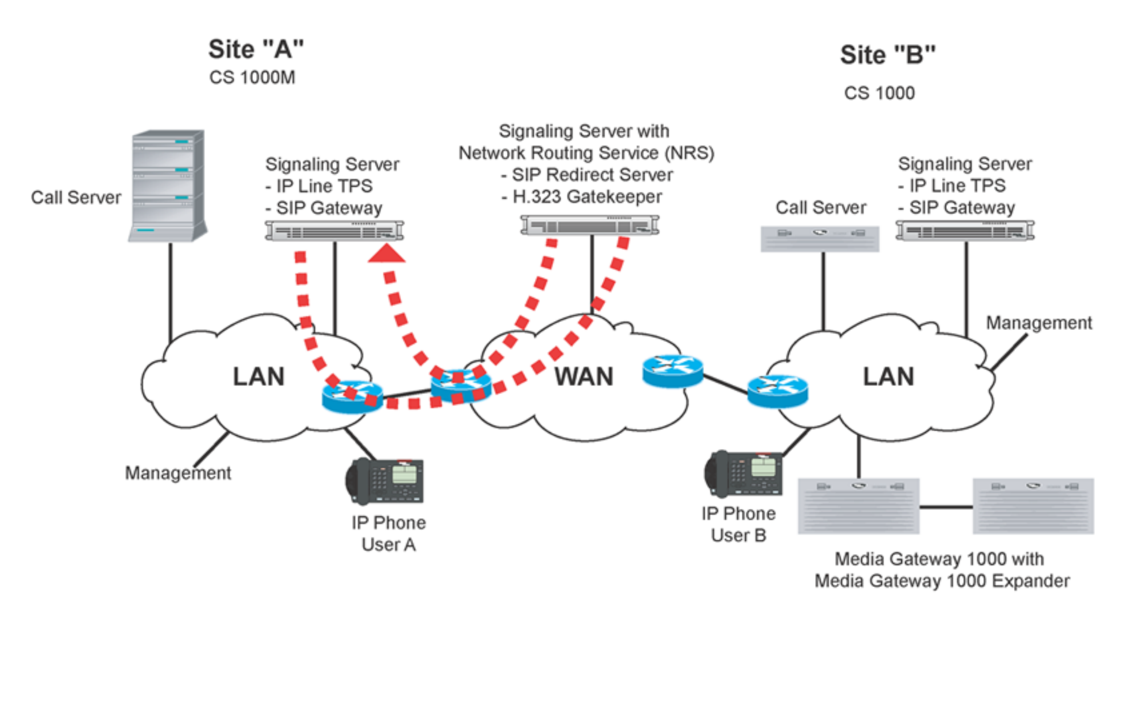
   Note: To select which Virtual Trunk to use for routing, Call Server A examines the number dialed, and uses various trunk routing and signaling features (for example, ESN and MCDN).

Figure 5
Call Server A routes the call to the IP network



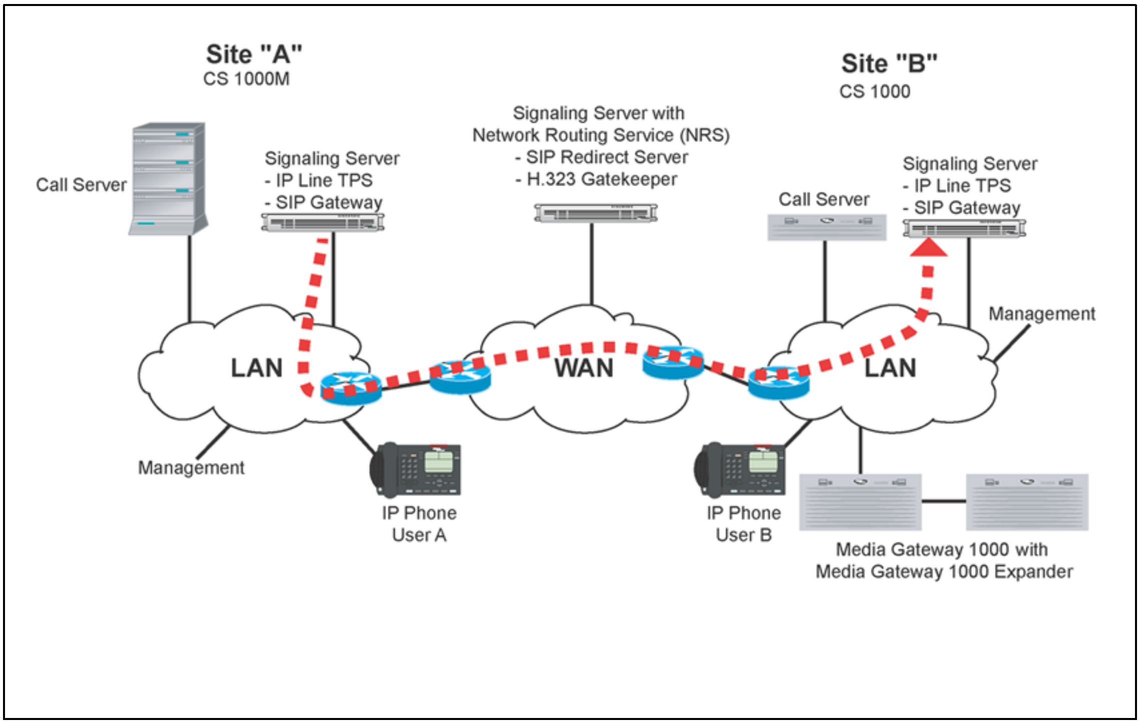
3. SIP Trunk Gateway A asks the NRS to search for the dialed DN in the database (for example, within the appropriate CDP domain). The NRS (SIP Redirect Server) sends the IP address of the SIP Trunk Gateway B to SIP Trunk Gateway A. See Figure 6 "The NRS sends the IP address of SIP Trunk Gateway B to SIP Trunk Gateway A" (page 49).

Figure 6
The NRS sends the IP address of SIP Trunk Gateway B to SIP Trunk Gateway A



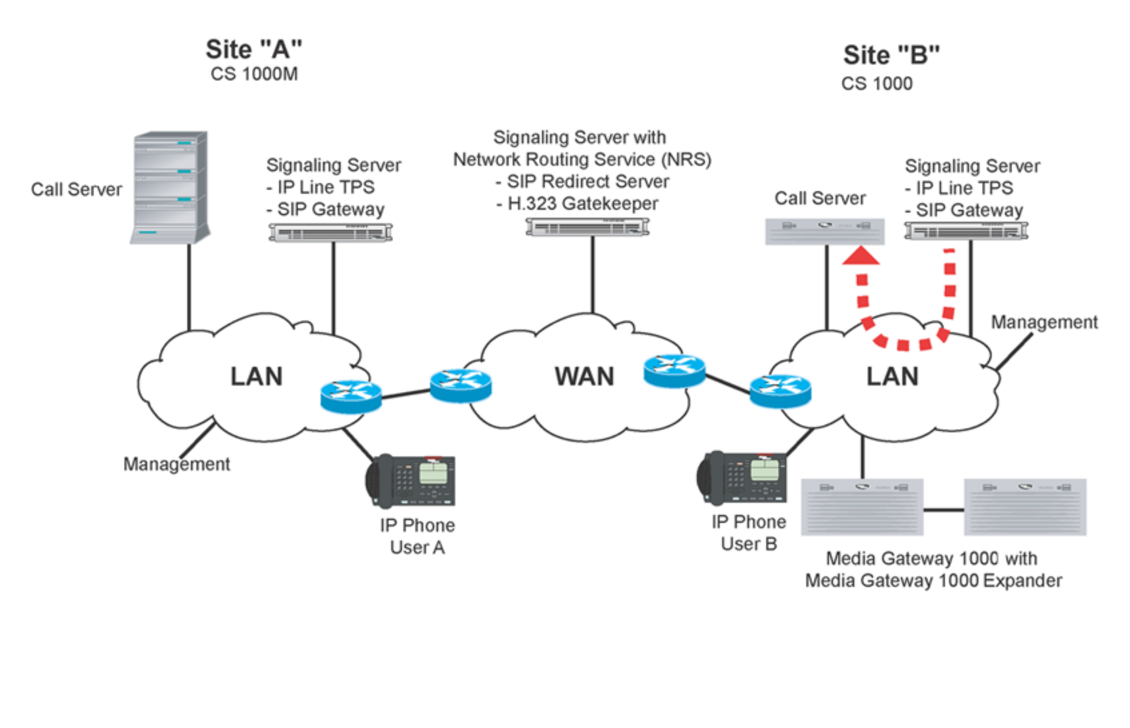
4. SIP Trunk Gateway A sends an INVITE message to SIP Trunk Gateway B, including the DN information. See Figure 7 "SIP Trunk Gateway A sends an INVITE message to SIP Trunk Gateway B" (page 50).

Figure 7
SIP Trunk Gateway A sends an INVITE message to SIP Trunk Gateway B



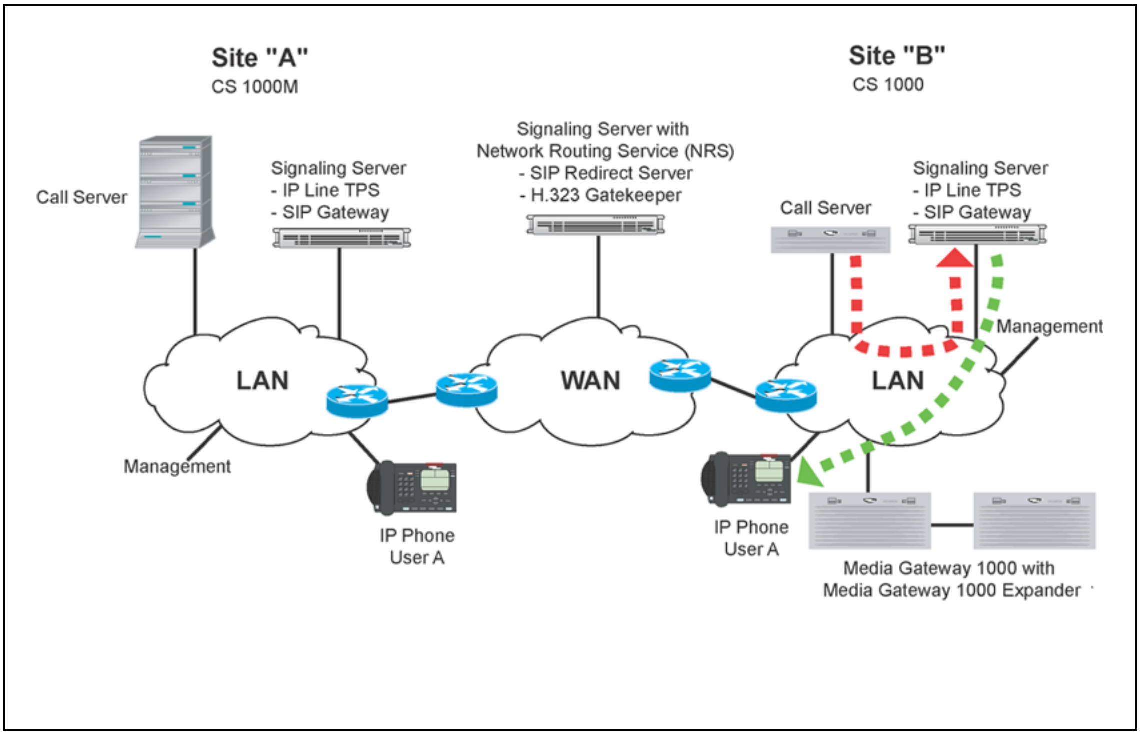
5. SIP Trunk Gateway B treats the incoming call from SIP Trunk Gateway A as an incoming Virtual Trunk call. SIP Trunk Gateway B sends the call to Call Server B over a Virtual Trunk. Call Server B also treats the call as an incoming call from a Virtual Trunk. See Figure 8 "SIP Trunk Gateway B sends the call to Call Server B over a Virtual Trunk" (page 51).

Figure 8
SIP Trunk Gateway B sends the call to Call Server B over a Virtual Trunk



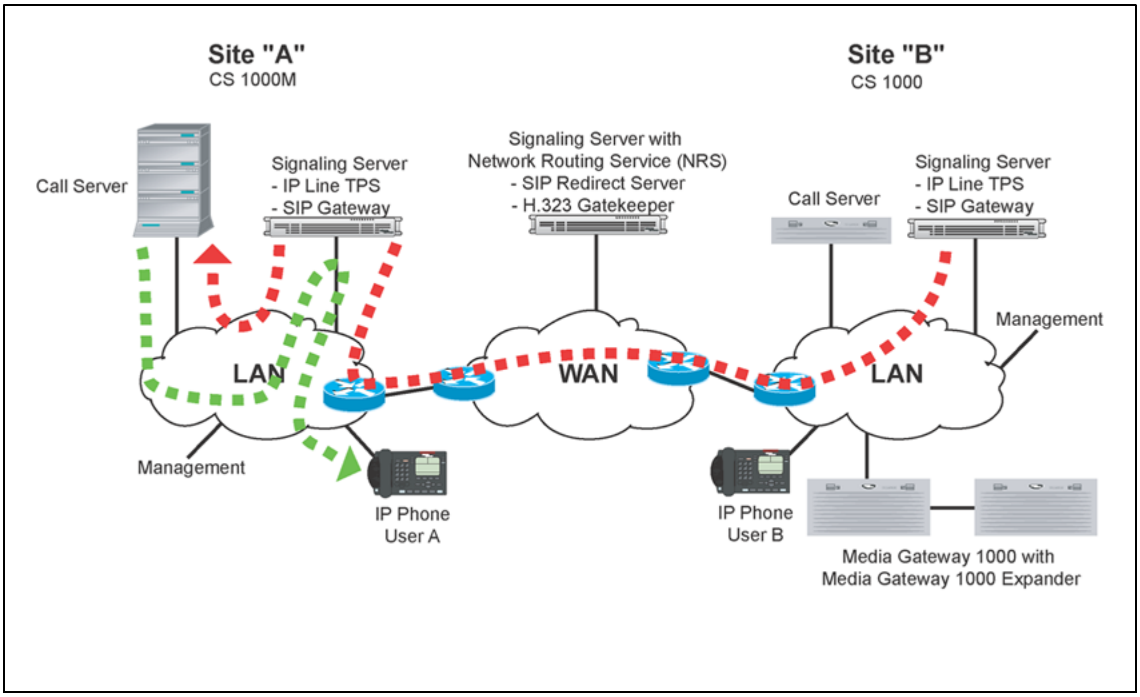
6. Call Server B selects the codec, allocates bandwidth, rings the telephone, and sends an ISDN Alert message to SIP Trunk Gateway B over the Virtual Trunk. See Figure 9 "Call Server B sends an Alert message to SIP Trunk Gateway B" (page 52).

Figure 9
Call Server B sends an Alert message to SIP Trunk Gateway B



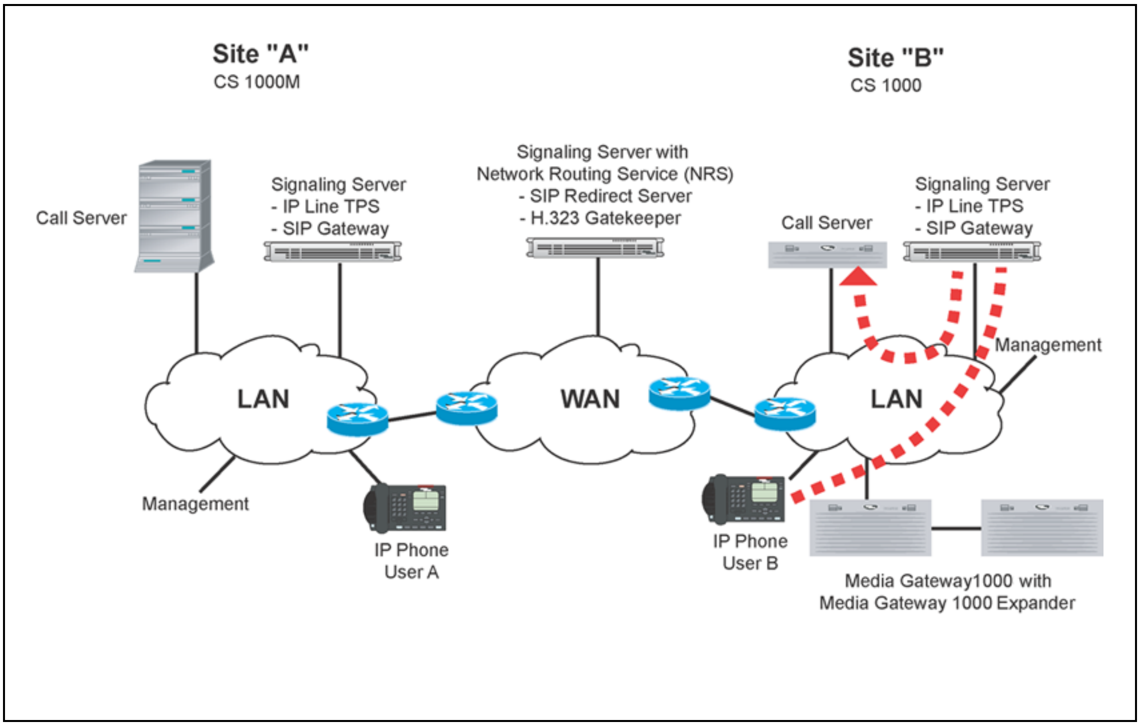
7. SIP Trunk Gateway B converts the ISDN Alert message to a SIP 180 response message. SIP Trunk Gateway B sends the SIP message to SIP Trunk Gateway A. SIP Trunk Gateway A converts the SIP 180 message back to the ISDN Alert message. SIP Trunk Gateway A then sends the message to Call Server A. Call Server A requests that the IP Phone play ringback tone. See Figure 10 "SIP Trunk Gateway B sends an Alert message to Call Server A" (page 53).

Figure 10
SIP Trunk Gateway B sends an Alert message to Call Server A



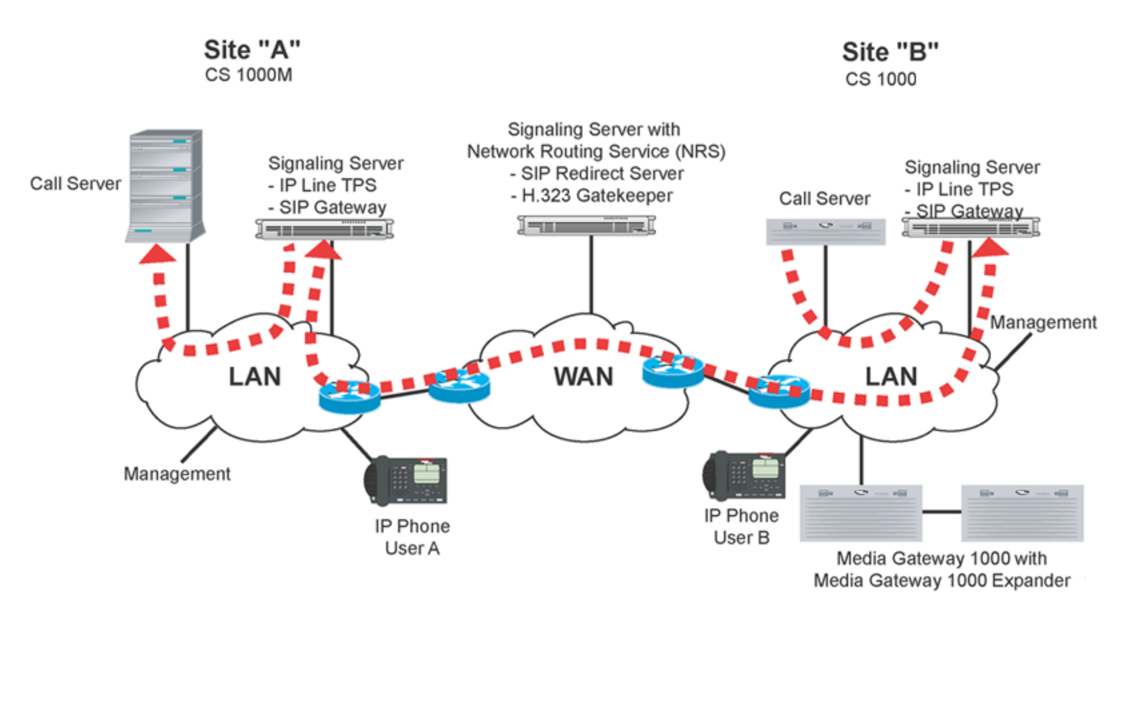
8. User B answers the call. A message is sent to Call Server B through the TPS on Signaling Server B. See Figure 11 "User B answers the call" (page 54).

Figure 11
User B answers the call



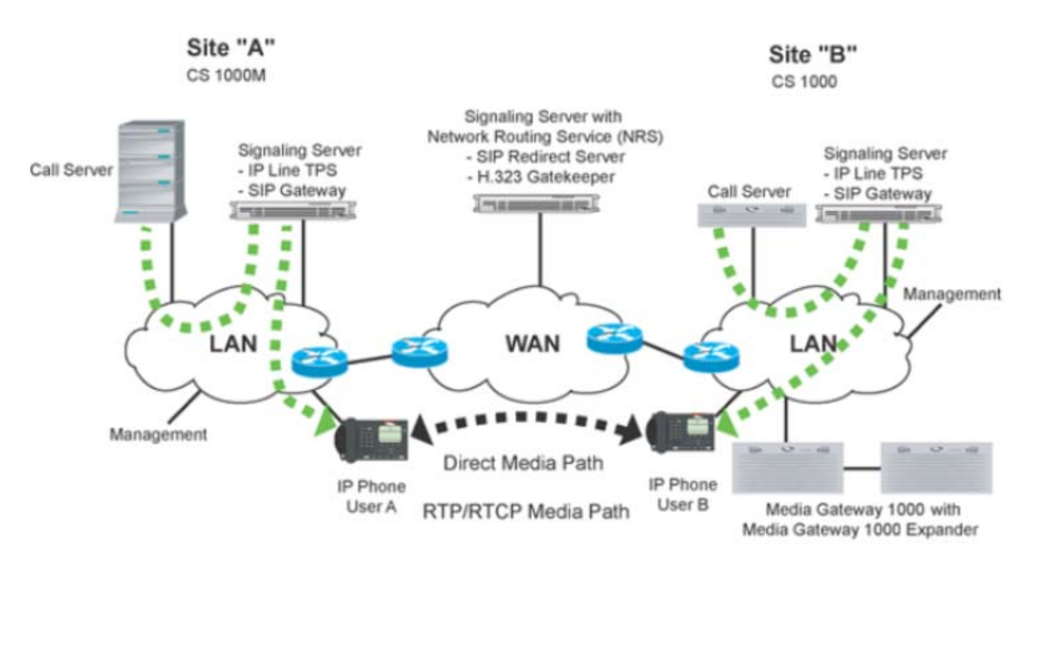
9. Call Server B sends an ISDN CONNECT message to SIP Trunk Gateway B. SIP Trunk Gateway B converts the CONNECT message to the SIP 200 OK message. SIP Trunk Gateway B sends the SIP 200 OK message to SIP Trunk Gateway A. SIP Trunk Gateway A sends an ACK message back to SIP Trunk Gateway B to acknowledge the SIP 200 OK message. SIP Trunk Gateway A converts the SIP 200 OK message back to the ISDN CONNECT message and sends the message to Call Server A over the Virtual Trunk. See Figure 12 "Call Server B sends an ACK message to SIP Trunk Gateway B" (page 55).

Figure 12
Call Server B sends an ACK message to SIP Trunk Gateway B



10. The Call Servers tell the IP Phones to start the direct IP media paths. The IP Phones then begin to transmit and receive voice over the IP network. See Figure 13 "IP Phones start the direct IP media paths" (page 56).

Figure 13
IP Phones start the direct IP media paths



Call scenarios

In the sections that follow, direct IP-media-path operation is described for a number of call scenarios. Each scenario uses IP Peer Networking to provide a direct IP media path between the peers taking part in the call. In all cases, the IP signaling path separates from the IP media path. Depending on the originating and terminating terminal types, the media path is between one of the following:

- IP Phone and IP Phone
- IP Phone and circuit-switched gateway
- circuit-switched gateway and circuit-switched gateway
- SIP Phone and SIP Phone (see *SIP Phone-to-SIP Phone communication)*
- SIP Gateway and SIP Phone (see SIP Trunk Gateway-to-SIP Phone communication)

In each case, the IP signaling path is the same; the trunk is virtual instead of physical.

IP Phone to circuit-switched telephone (on separate Call Servers)

An IP Phone on Node A calls a circuit-switched telephone (for example, an analog [500/2500-type] telephone) on Node B.

The Call Server on the originating node selects an ISDN route and Virtual Trunk, based on the dialed digits translation. The ISDN signaling routes through the Signaling Server and encodes using SIP.

On the terminating node, the SIP signaling is received at the Signaling Server, and converts the SIP message to an ISDN message. The ISDN message is forwarded to the Call Server. The terminating Call Server translates the received digits to the DN of a circuit-switched device. The Call Server determines that the call is incoming on a Virtual Trunk and terminating on a circuit-switched device, and selects a DSP resource on a Voice Gateway Media Card. The DSP performs IP-to-circuit-switched conversion when the call is established.

When the terminating circuit-switched party answers the call, the terminating Call Server returns an ISDN CONNECT message. The message is sent to the SIP Trunk Gateway on the Signaling Server. The SIP Trunk Gateway on the Signaling Servers converts the ISDN CONNECT message to a SIP 200 OK message and the Signaling Server completes the exchange of IP media information required to establish the IP media path. The originating and terminating Call Servers establish a direct two-way IP media path between the IP Phone and the DSP. The terminating node also establishes a circuit-switched speechpath between the DSP and the circuit-switched telephone.

> ***Note:*** If a Voice Gateway Media Card channel is not available when required for IP to circuit-switched connections, call processing treats the scenario the same way current traffic timeslot blocking is handled. If all Virtual Trunks in a route are busy when call routing is attempted, the routing operates the same way as physical trunks by routing the call to the next available route selection.

IP Phone to Recorded Announcement or Music

In certain call scenarios, an IP Phone requires a Recorded Announcement (RAN) or Music treatment from a remote node. For example, an IP Phone is placed on hold by a party on a remote node that has Music on Hold configured.

When the IP Phone is placed on hold by the holding party, the direct IP media path that had been established between the two parties is torn down. A new IP media path is established between the IP Phone and a circuit-switched gateway on the node providing the Music.

The media path, in this case, is one way only (from the circuit-switched gateway to the IP Phone). This media-path redirection is initiated by the node providing the Music, using the SIP re-INVITE or UPDATE methods. No ISDN signaling is exchanged between the nodes, and the call state on the originating node is unchanged.

IP Peer Networking supports RAN Broadcast and Music Broadcast. The RAN and Music Broadcast features enable multiple listeners to share the same RAN and Music trunks to listen to a recorded announcement or music. However, one DSP channel is required for each user. IP Peer Networking does not support IP broadcast/multicast of RAN or Music.

When the holding party retrieves the held call, the media path is torn down, and a two-way IP media path is reestablished between the parties.

Virtual Trunk to Virtual Trunk

An incoming call to a node over a Virtual Trunk is routed over another Virtual Trunk based on the translation of digits in the SIP INVITE message. A call between two parties on remote nodes is tandemed through this node.

The call originates on the incoming Virtual Trunk. ISDN signaling is converted and exchanged between the originating node and the tandem node using SIP. The call terminates on the outgoing Virtual Trunk, and ISDN signaling is converted and exchanged between the tandem node and the terminating node using SIP.

The ISDN signaling generated at the end node is sent through the tandem node and processed by the Call Server. The Call Server processes the call as it does a normal tandem call. The exchange of IP call parameters between the end nodes is sent through the tandem node’s Signaling Server and Call Server, so each end node can establish a direct IP media path between end parties.

The IP media path is established directly between the originating and terminating parties on the end nodes. No media resources are used on the tandem switch. When trunks are not optimized, signaling continues to be handled in a tandem manner, even though the media path is direct.

Tandem operations

All media paths route directly over IP networks. However, to maintain proper control points and billing records for a call, sometimes signaling must be indirect. The following sections describe indirect signaling operations for these scenarios.

Direct tandem calls Because SIP IP Peer Networking uses the NRS (specifically the SIP Redirect Server) for address resolution, there is minimal requirement for tandem calls. With an NRS (SIP Redirect Server), each node can obtain the IP address of the terminating node. Therefore, calls route directly to the terminating node and not through a tandem node.

Feature modification (for example, Call Transfer) can cause calls to tandem. Tandem calls also occur when routing is configured as tandem, so accounting records can generate during calls from a third-party gateway.

Tandem feature calls When a tandem call occurs due to a transfer operation, the IP media paths between the originating party and the "transferred-to" party must be redirected to each other. This redirection is initiated by the transferring (tandem) node.

This scenario describes a form of the Trunk Route Optimization (TRO)/MCDN feature.

When a tandem call occurs due to a Call Forward operation, it attempts to use TRO to optimize the route between the originating and "transferred-to" parties. In the event packaging or user provisioning selections mean that TRO is not supported, the tandem node initiates media path redirection for both parties.

TRO is used when a call from Node A to Node B forwards to Node C. Node B sends a TRO facility message to Node A. The message contains the digits of the "forwarded-to" party. Node A resolves these digits to a route and determines whether it has a direct route configured to Node C.

IP Peer handling of TRO differs slightly from the PRI handling at this point. With PRI, each destination has a dedicated route and ISDN link. With IP Peer, in the Node A routing configuration, all remote locations are reached using the same Virtual Trunk (the SIP Redirect Server subsequently translates the digits to separate IP nodes). When TRO is attempted at Node A, the call processing finds that the new destination is accessed through the same Virtual Trunk route, and accepts the TRO even though the call does not have an alternate direct route to Node C. The tandem call routing through Node B is cleared. Node A places a new call through the same Virtual Trunk route and IP D-channel that was used for the original call to Node B. The SIP Redirect Server translation identifies the correct destination, Node C, and the call is placed directly to that node.

In cases where the TRO feature does not optimize trunks, the Virtual Trunks must remain busy at Nodes A, B, and C until the call is released. A direct media path between Node A and Node C supports the connection; Node B is not on the media path. This eliminates voice quality problems caused by multiple transcoding steps.

TRO versus TAT on transferred call The TRO feature will optimize a redirected call initiated on the transfer key. Station A calls Station B on Node 1. Station B puts the call on hold and initiates a call transfer over a SIP trunk to Station C on Node 2, which call forwards no answer to Station D on Node 1. While Station D is ringing (Station B hasn’t completed the call transfer), optimization will be done by the TRO feature. If Station B completes the call transfer while Station C is ringing, and then the call is forwarded no answer to Station D, the TRO feature will optimize the redirected call and release the two SIP trunks connecting Node 1 and Node 2 before Station D answers the call.

Circuit-switched tandem calls The IP Peer Networking feature supports circuit-switched tandem calls by configuring a circuit-switched TIE trunk on a CS 1000 system or gateway which routes calls across the IP network. The signaling over the circuit-switched trunk can use any of the TIE trunks supported in traditional MCDN circuit-switched networks.

Virtual Trunk calls in conference

A party on Node A calls a party on Node B. The party on Node B creates a three-party conference with a party on Node C. A circuit-switched conference circuit is used on Node B. Each party has their media path redirected to a separate circuit-switched gateway on Node B. Circuit-switched speech paths are established between each circuit-switched gateway and the conference bridge.

Virtual Trunk to circuit-switched party transferred to an IP Phone

The following occurs in this scenario:

- A call is established between a party on a remote node (the caller) and a circuit-switched party on the local node (the called party) using a Virtual Trunk.

  A media path exists between the caller (which can be an IP Phone or a circuit-switched gateway) and a circuit-switched gateway on the local node.

- The called party transfers the call to an IP Phone on the local node.

  When the called party initiates the transfer operation, the caller is placed on hold, using the re-INVITE message. The existing media path remains allocated. A local call (called a consultation call) is set up between the transferring called party and the local IP Phone to which the call is to be transferred.

- When the transfer is complete, the consultation call is released, and a call is set up between the caller and the IP Phone to which the call was transferred. The original media path between the caller and the called party is redirected using the SIP re-INVITE or UPDATE methods. Because the IP Phone to which the call was transferred is not a circuit-switched telephone, the circuit-switched gateway resource is released.

- A direct media path is set up between the caller and the IP Phone.

Virtual Trunk to a circuit-switched party transferred before answer to an IP Phone

The following occurs in this scenario:

- A call is established between a party on a remote node (the caller) and a circuit-switched party on the local node (the called party) over a Virtual Trunk.

  A direct IP media path exists between the caller (for example, an IP Phone or circuit-switched gateway) and a circuit-switched gateway on the local node.

- The called party initiates a transfer to an IP Phone on the local node before answering the call. While the IP Phone is ringing, the called party completes the transfer by disconnecting or pressing the Transfer key. The caller receives ringback tone.

  When the called party initiates the Transfer operation, the incoming Virtual Trunk (and indirectly, the caller) is placed on hold, and the direct IP media path between the caller and the circuit-switched gateway is torn down. If Music or RAN is configured, a new IP media path is established between a circuit-switched gateway and the caller.

- When the called party completes the "transfer before answer", ringback tone is provided to the caller. A new one-way IP media path is established between a circuit-switched gateway on the node providing the ringback tone and the caller. The node providing the ringback tone initiates this media path "redirection" using the SIP re-INVITE or UPDATE methods. It does not use ISDN signaling for this purpose.

- When the party on the transferred-to IP Phone answers, another media path redirection occurs. The media path between the circuit-switched gateway and the caller is released, and a new two-way IP media path is established between the caller and the party answering the IP Phone to which the call was transferred. This uses the SIP re-INVITE or UPDATE methods.

IP Phone to local IP Phone transferred to a Virtual Trunk

A call is established between two IP Phones on the same node. A direct media path exists between the two telephones. One of the parties initiates a transfer to a party on a remote node.

When the IP Phone party initiates the transfer, call processing on the local node places the other party on hold. The media path between the two IP Phones is torn down. A call is set up between the transferring IP Phone and the remote party (this could be an IP Phone or circuit-switched telephone). See "IP Phone to IP Phone (on separate Call Servers)" (page 46).

When the transferring IP Phone completes the transfer before answer, the consultation call between the IP Phone and the remote party is torn down and a call is set up between the transferred IP Phone and the remote party. The media path that existed between the remote party and the transferring IP Phone is redirected using the SIP re-INVITE or UPDATE methods. No ISDN signaling is exchanged between the nodes, and the call state on the terminating node is unchanged. A direct IP media path is established between the transferred IP Phone and the remote party.

H.323 signaling

Contents

This section contains information on the following topics:

Direct IP Media Paths

With IP Peer Networking, the H.323 Gateway Signaling software enables direct IP voice paths to IP devices. An endpoint is the H.323 Gateway that terminates an H.323 signaling stream. An H.323 Gateway that terminates H.323 signaling registers at an H.323 Gatekeeper as an endpoint. IP Phones interact with the Gateway software to appear as H.323 devices that support Direct IP Media Paths.

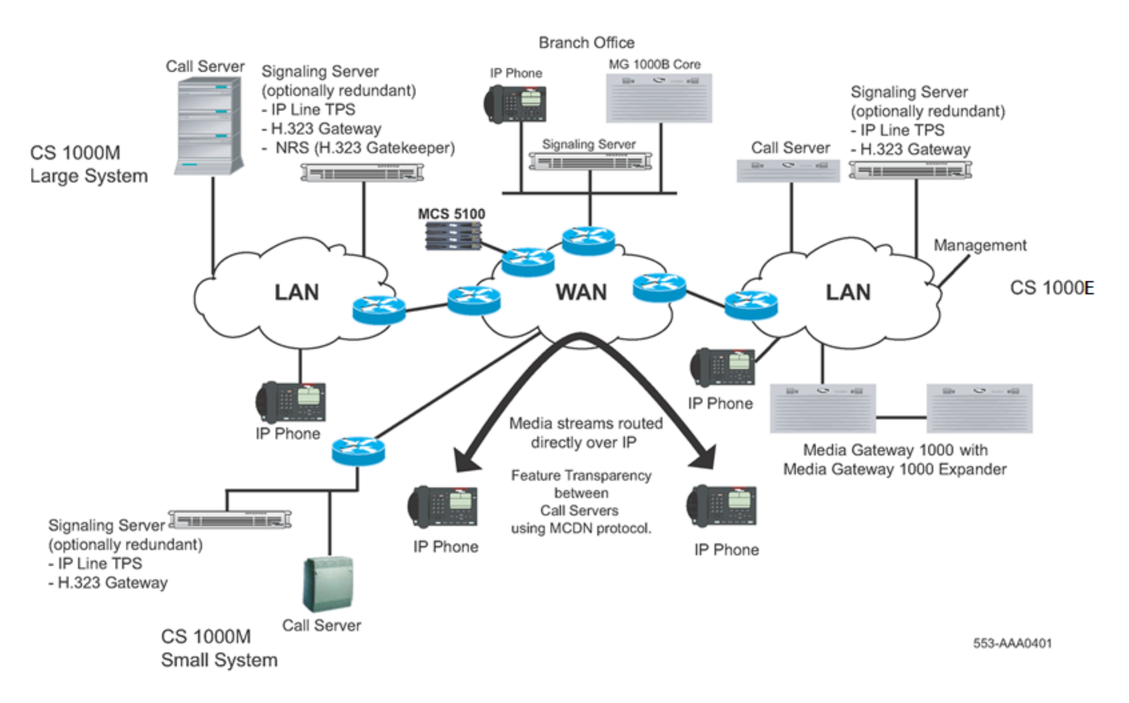
> ***Note 1:*** IP Peer Networking supports both Media Gateways and third-party Gateways that have been tested for compatibility. Use the Gateway to enable communication between an H.323 or SIP network and circuit-switched equipment. Interfaces provided by Media Gateways operate in H.323/SIP standard mode and support MCDN feature capabilities. They operate autonomously in the network.
>
> ***Note 2:*** A Media Gateway is a gateway that uses a protocol similar to the Media Gateway Control Protocol (MGCP). The Media Gateway houses peripheral cards. Media Gateways are controlled directly by the Call Server.

Direct IP Media Path functionality ensures that, when any IP device in the network (for example, an IP Phone) connects to another IP address (for example, an IP Phone), the media path uses direct IP connections and does not pass through a central circuit-switched PBX. When the connection is made between a Virtual Trunk and a circuit-switched device (for example, a PRI trunk), a DSP resource is allocated to transcode the media stream from IP to circuit-switched.

When the network address of the local IP device or DSP resource is determined, the address is signaled using standard H.323 protocol to the far end so a direct media path can be established. If a call modification operation is involved (for example, Call Transfer), further signaling of the address information occurs using standard H.323 Pause and Reroute protocol.

Figure 14 "An example of IP Peer Networking using Virtual Trunk and direct media paths" (page 64) shows a media path routed directly over IP, not using a circuit switch.

Figure 14
An example of IP Peer Networking using Virtual Trunk and direct media paths



IP Phone to IP Phone (on separate Call Servers)

An IP Phone at Site A calls an IP Phone at Site B (see Figure 15 "User A dials User B" (page 66)). When the user presses a key on the IP Phone, a signaling message is carried over the IP network.

The Call Server on the originating node selects an ISDN route and a virtual IP trunk, based on the dialed digits translation. After terminating on a Virtual Trunk, D-channel signaling occurs over IP. This includes basic call setup signals (Q.931 over IP, as well as Nortel MCDN signaling over IP, which is used for networking features). The ISDN Q.931 signaling is routed using the

Signaling Server and encoded using the H.323 protocol. MCDN messages are carried within the H.323 protocol, using standard H.323 facilities for proprietary extensions.

On the terminating node, the H.323 signaling is received at the Signaling Server, and the ISDN Q.931 messages are forwarded to the Call Server. The terminating Call Server translates the received digits to an IP Phone DN. When the terminating IP Phone answers the call, the terminating node returns a Q.931 CONNECT message, and the Signaling Servers complete the exchange of the IP media information required to establish the IP media path. The originating and terminating Call Servers establish a direct two-way IP media path between the two IP Phones.

Basic network call walk-through

When a user makes a call on a CS 1000 system, the dialed digits are translated to determine if the user is attempting to reach an internal or external telephone.

By default, H.323 on CS 1000 systems uses en bloc signaling. For overlap signaling, refer to "Overlap signaling" (page 251).

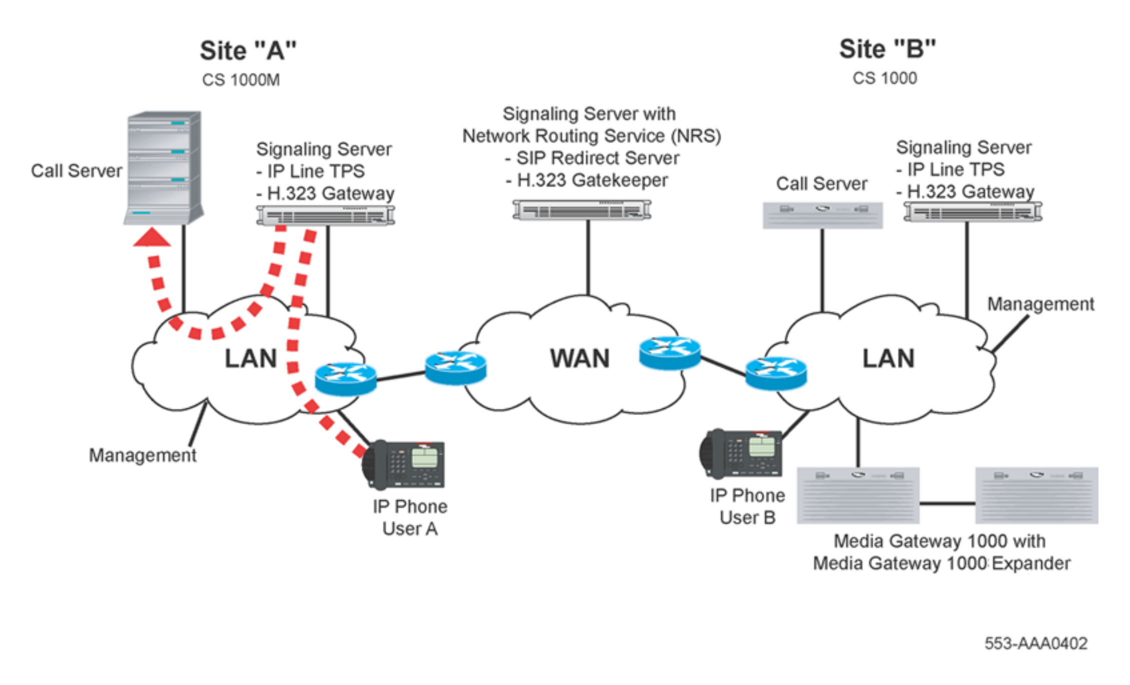
If the user is attempting to reach an internal telephone, the call is terminated on the internal device. When the system determines that the user is attempting to reach a telephone or service using the IP network, the call routes to the H.323 Gateway software. The H.323 Gateway software uses the NRS (specifically the H.323 Gatekeeper) to help with call routing.

> ***Note 1:*** Configure Virtual Trunk routes as circuit-switched routes. Use CS 1000 Element Manager or LD 14 and LD 16 in the Command Line Interface (CLI). "Configuring the Virtual routes and trunks" (page 189)
>
> ***Note 2:*** Only the primary messages are illustrated in the following call flows.

The following scenario describes the Direct IP Media Path functionality for a basic network call using en bloc signaling:

1. User A on Call Server A dials the DN of User B on Call Server B. Call Server A collects the digits through the Terminal Proxy Server (TPS) on the Signaling Server A. See Figure 15 "User A dials User B" (page 66).

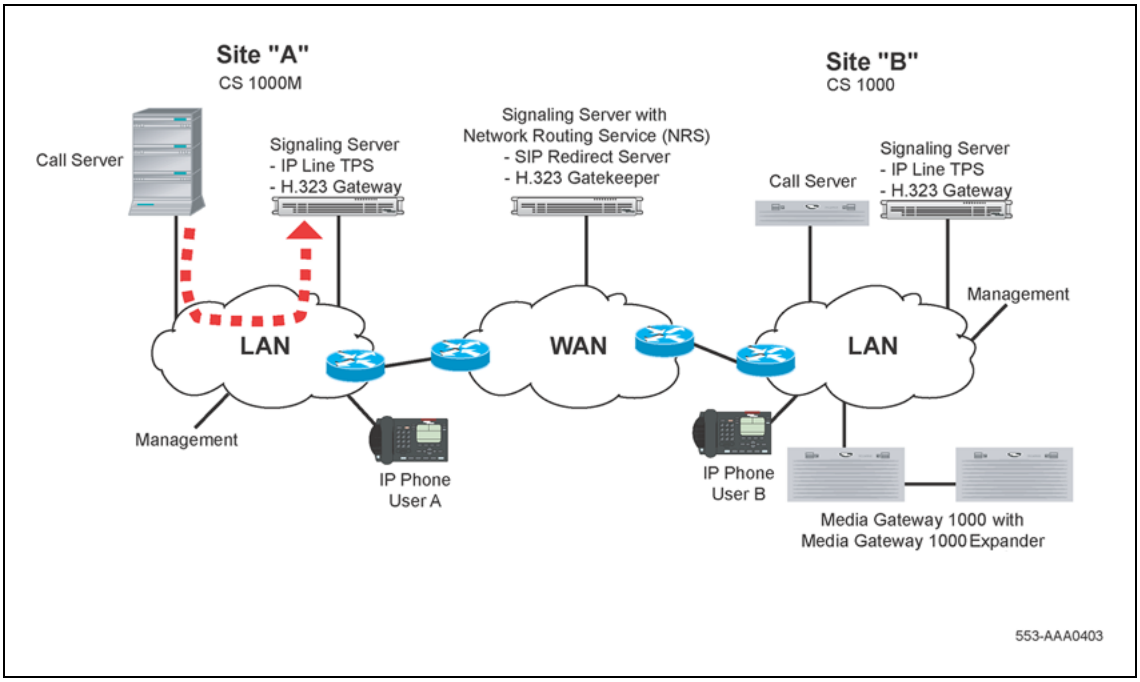
Figure 15
User A dials User B



2. Call Server A determines that the dialed DN is at another site. Call Server A selects the codec list, allocates bandwidth, and routes the call to the IP network using a Virtual Trunk and an H.323 Gateway. See Figure 16 "Call Server A routes the call to the IP network" (page 67).

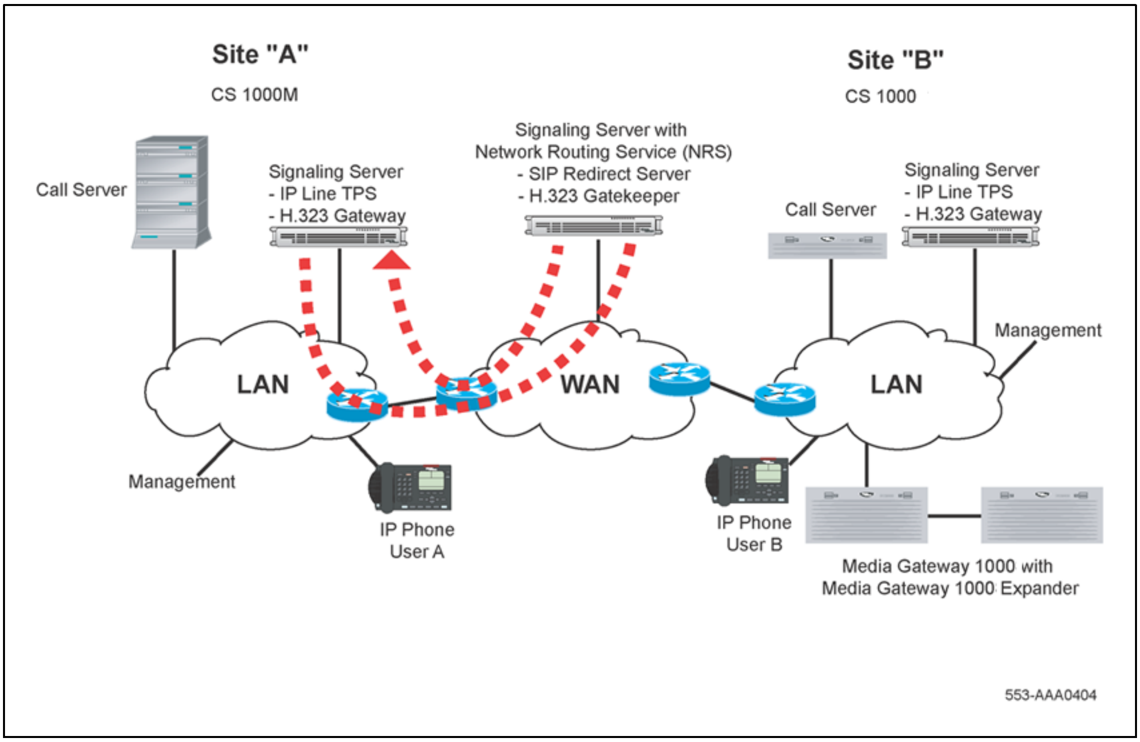
   Note: To select which Virtual Trunk to use for routing, Call Server A examines the number dialed and uses various trunk routing and signaling features (for example, ESN and MCDN).

Figure 16
Call Server A routes the call to the IP network



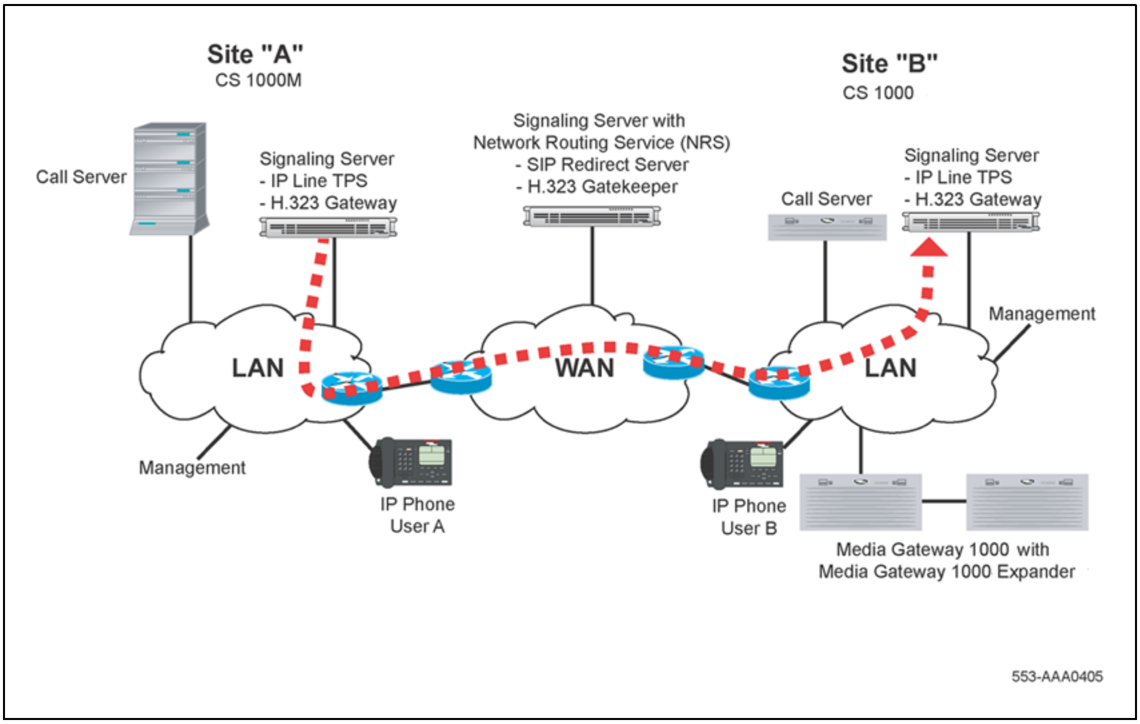
3. H.323 Gateway A asks the NRS (specifically the H.323 Gatekeeper) to search for the dialed DN in the database (for example, within the appropriate CDP domain). The NRS (H.323 Gatekeeper) sends the IP address of H.323 Gateway B to H.323 Gateway A. See Figure 17 "The H.323 Gatekeeper sends the IP address of H.323 Gateway B to H.323 Gateway A" (page 68).

Figure 17
The H.323 Gatekeeper sends the IP address of H.323 Gateway B to H.323 Gateway A



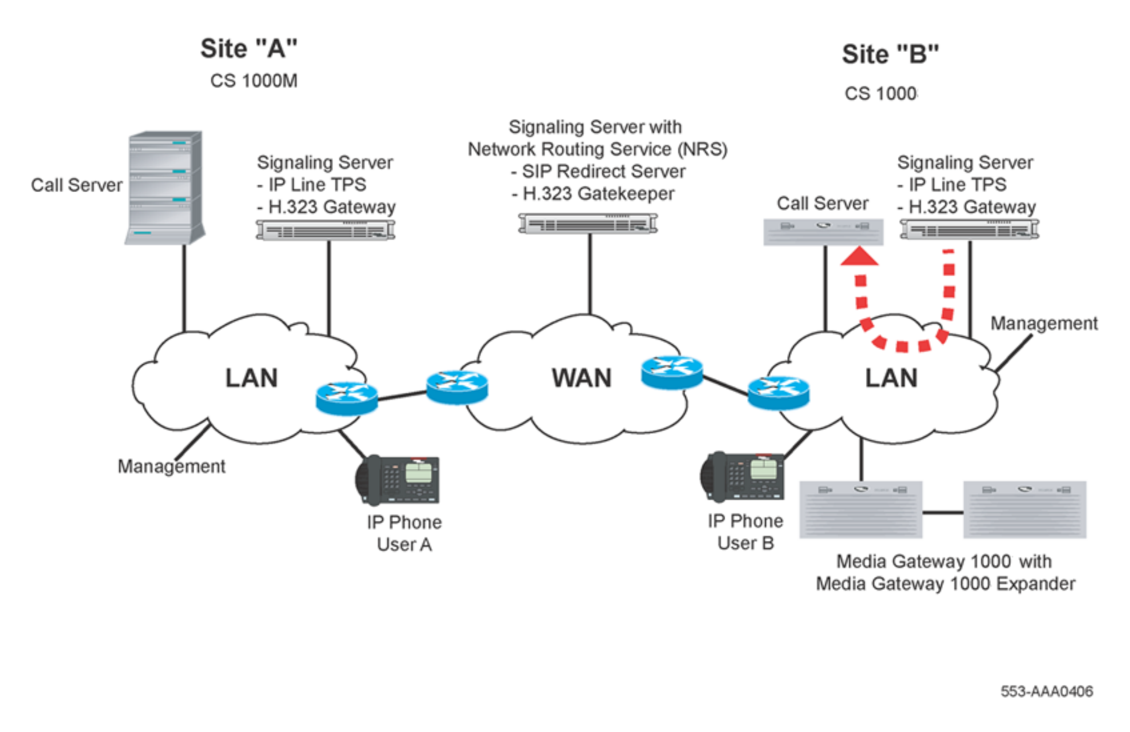
4. H.323 Gateway A sends a SETUP message to H.323 Gateway B, including the DN information. See Figure 18 "H.323 Gateway A sends a SETUP message to H.323 Gateway B" (page 69).

Figure 18
H.323 Gateway A sends a SETUP message to H.323 Gateway B



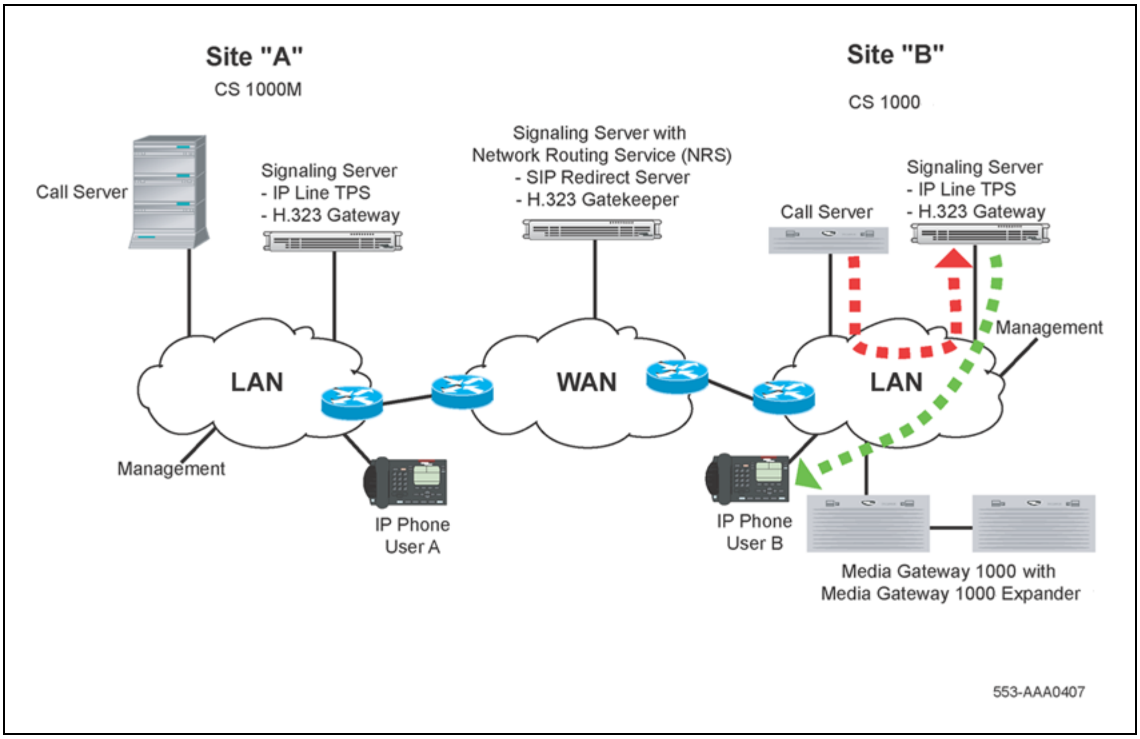
5. H.323 Gateway B treats the call as an incoming call from a Virtual Trunk. H.323 Gateway B sends the call to Call Server B over a Virtual Trunk. Call Server B also treats the call as an incoming call from a Virtual Trunk. See Figure 19 "Gateway B sends the call to Call Server B over a Virtual Trunk" (page 70).

Figure 19
Gateway B sends the call to Call Server B over a Virtual Trunk



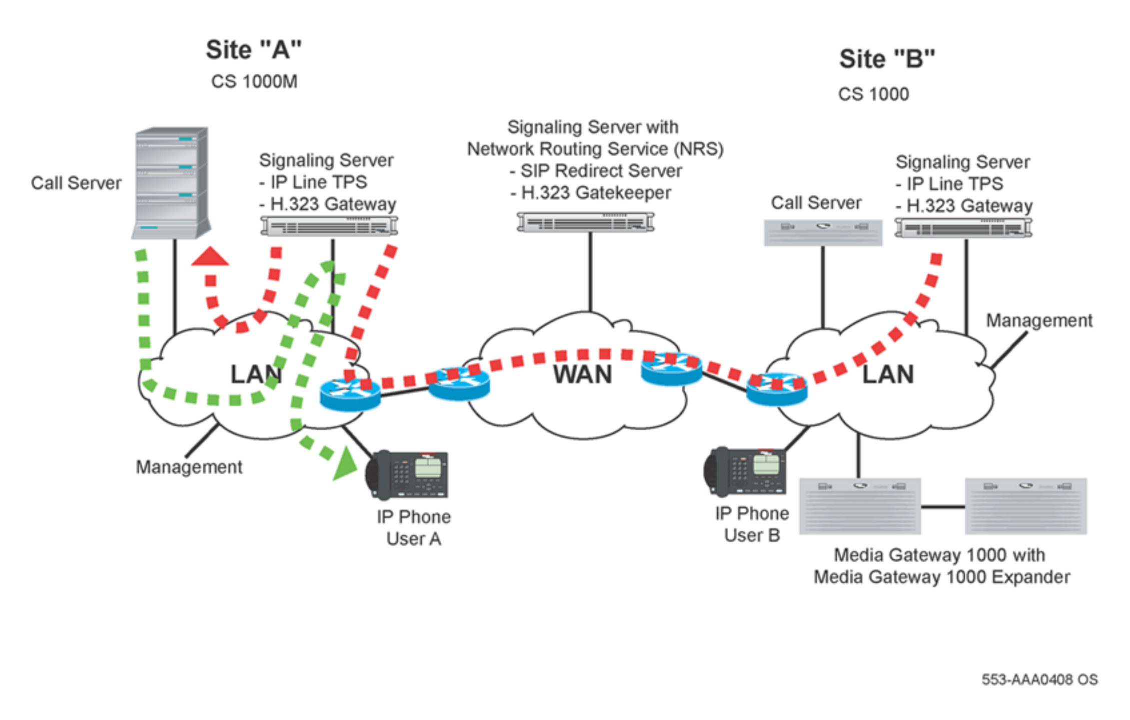
6. Call Server B selects the codec, allocates bandwidth, rings the telephone, and sends an alerting message to H.323 Gateway B. See Figure 20 "Call Server B sends an alerting message to H.323 Gateway B" (page 71).

Figure 20
Call Server B sends an alerting message to H.323 Gateway B



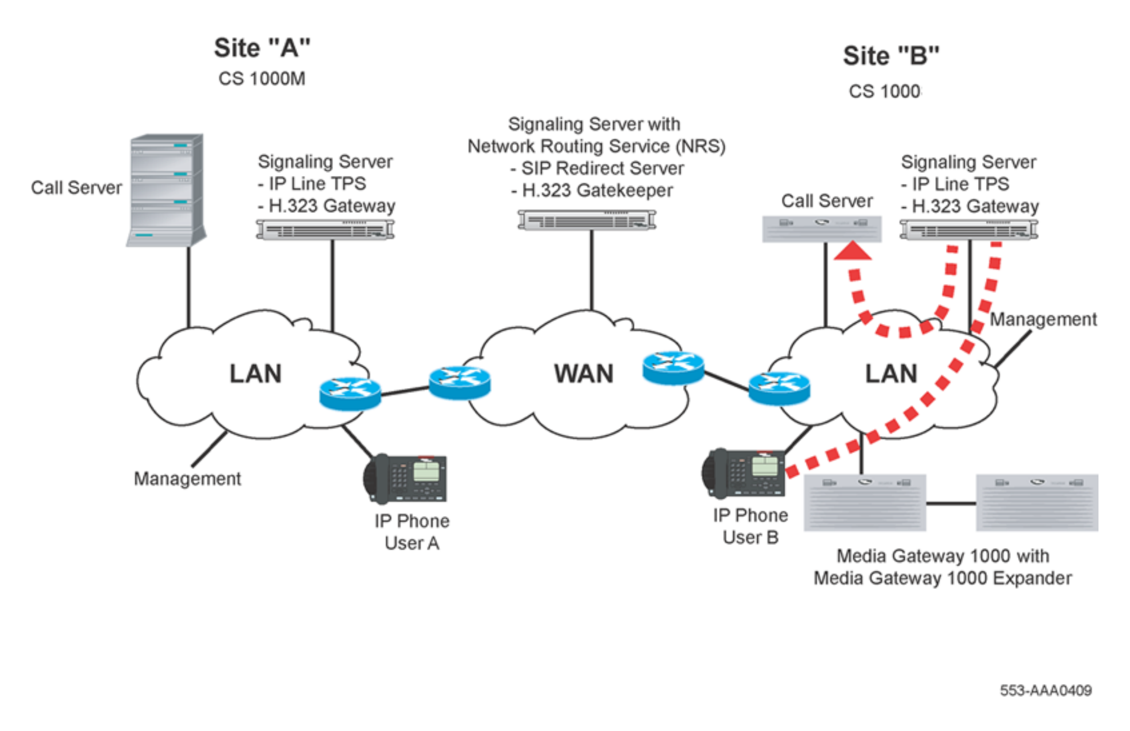
7. H.323 Gateway B sends an alerting message to Call Server A. Call Server A requests that the IP Phone play ringback tone. See Figure 21 "H.323 Gateway B sends an alerting message to Call Server A" (page 72).

Figure 21
H.323 Gateway B sends an alerting message to Call Server A



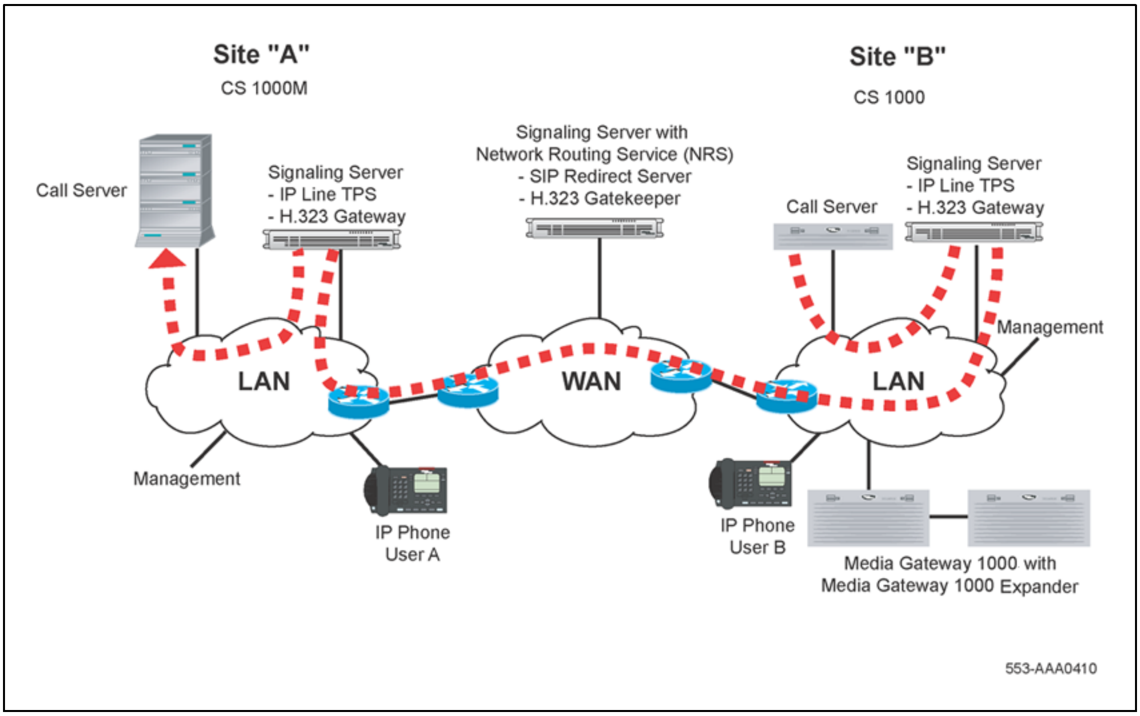
8. User B answers the call. A message is sent to Call Server B through the TPS on the Signaling Server. See Figure 22 "User B answers the call" (page 73).

Figure 22
User B answers the call



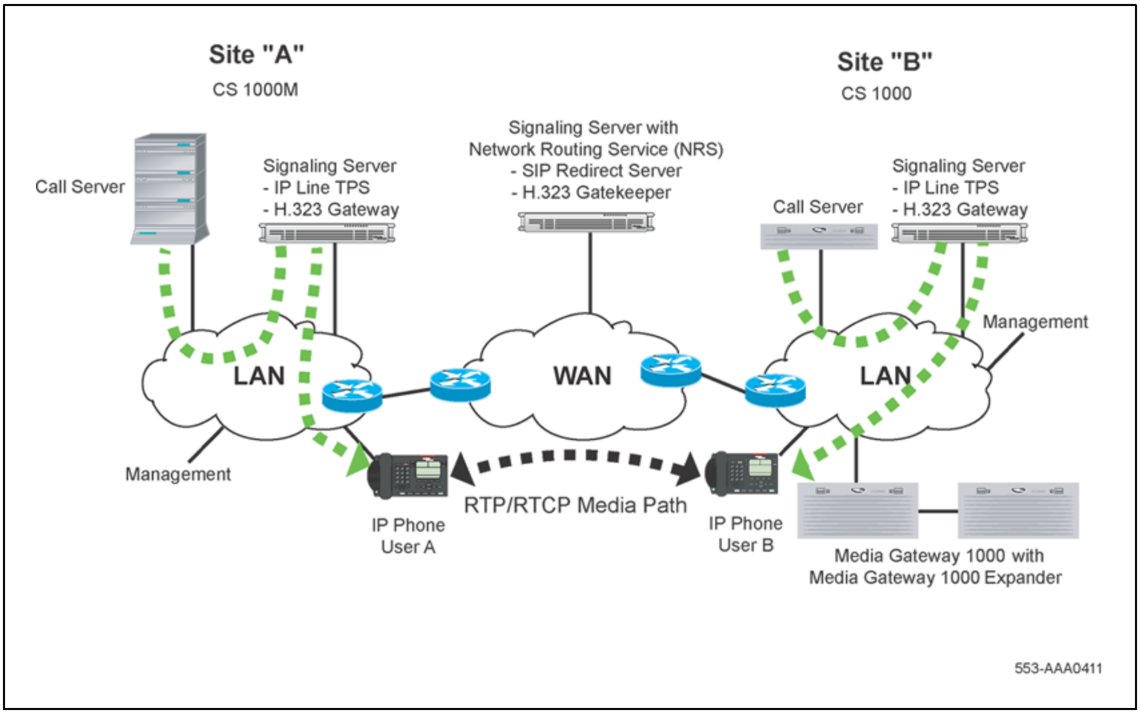
9. Call Server B sends a CONNECT message to H.323 Gateway B. H.323 Gateway B sends an H.323 CONNECT message to H.323 Gateway A and Call Server A. See Figure 23 "Call Server B sends a CONNECT message to Gateway B" (page 74).

Figure 23
Call Server B sends a CONNECT message to Gateway B



10. The Call Servers tell the IP Phones to start the direct IP media paths. The IP Phones then begin to transmit and receive voice over the IP network. See Figure 24 "IP Phones start the direct IP media paths" (page 75).

Figure 24
IP Phones start the direct IP media paths



Call scenarios

In the sections that follow, direct IP media path operation is described for a number of call scenarios. Each scenario uses IP Peer Networking to provide a direct IP media path between the peers taking part in the call. In all cases, the IP signaling path separates from the IP media path. Depending on the originating and terminating terminal types, the media path is between one of the following:

- IP Phone and IP Phone
- IP Phone and circuit-switched gateway
- circuit-switched gateway and circuit-switched gateway

In each case, the IP signaling path is the same; the trunk is virtual instead of physical.

IP Phone to circuit-switched telephone (on separate Call Servers)

An IP Phone on Node A calls a circuit-switched telephone (for example, an analog [500/2500-type] telephone) on Node B.

The Call Server on the originating node selects an ISDN route and Virtual Trunk, based on the dialed digits translation. The ISDN Q.931 signaling messages that route through the Signaling Server are encoded using the H.323 protocol.

On the terminating node, the H.323 signaling is received at the Signaling Server, and the ISDN Q.931 messages forward to the Call Server. The terminating Call Server translates the received digits to the DN of a circuit-switched device. The Call Server determines that the call is incoming on a Virtual Trunk and terminating on a circuit-switched device, and selects a DSP resource on a Voice Gateway Media Card. The DSP performs IP-to-circuit-switched conversion when the call is established.

When the terminating circuit-switched party answers the call, the terminating node returns a Q.931 CONNECT message, and the Signaling Servers complete the exchange of IP media information required to establish the IP media path. The originating and terminating Call Servers and Media Gateway SSC establish a direct two-way IP media path between the IP Phone and the DSP. The terminating node also establishes a circuit-switched speechpath between the DSP and the circuit-switched telephone.

> ***Note:*** If a Voice Gateway Media Card channel is not available when required for IP to circuit-switched connections, call processing treats the scenario the same way current call blocking is handled. If all Virtual Trunks in a route are busy when call routing is attempted, the routing operates the same way as physical trunks by routing the call to the next available route selection.

IP Phone to Recorded Announcement or Music

In certain call scenarios, an IP Phone requires a Recorded Announcement (RAN) or Music treatment from a remote node. Such a scenario could occur, for example, if an IP Phone is placed on hold by a party on a remote node that has Music on Hold configured.

When the IP Phone is placed on hold by the holding party, the direct IP media path that had been established between the two parties is torn down. A new IP media path is established between a circuit-switched gateway on the node providing the Music and the IP Phone.

The media path, in this case, is one way only (from the circuit-switched gateway to the IP Phone). This media path redirection is initiated by the node providing the Music, using the H.323 third-party initiated pause and re-routing mechanism. No ISDN Q.931 signaling is exchanged between the nodes, and the call state on the originating node is unchanged.

IP Peer Networking supports RAN Broadcast and Music Broadcast. The RAN and Music Broadcast features enable multiple listeners to share the same RAN and Music trunks to listen to a recorded announcement or music. However, one DSP channel is required for each user. IP Peer Networking does not support IP broadcast/multicast of RAN or Music.

When the holding party retrieves the held call, the media path is torn down, and a two-way IP media path is reestablished between the parties.

Virtual Trunk to Virtual Trunk

An incoming call to a node over a Virtual Trunk is routed over another Virtual Trunk based on the translation of digits in the Q.931 SETUP message. A call between two parties on remote nodes is tandemed through this node.

The call originates on the incoming Virtual Trunk. ISDN Q.931 signaling is exchanged between the originating node and the tandem node using the H.323 protocol. The call terminates on the outgoing Virtual Trunk, and ISDN Q.931 signaling is exchanged between the tandem node and the terminating node using the H.323 protocol.

The ISDN Q.931 signaling generated at the end node is sent through the tandem node and processed by the Call Server. The Call Server processes the call as it does a normal tandem call. The exchange of IP call parameters between the end nodes is sent through the tandem node’s Signaling Server and Call Server, so each end node can establish a direct IP media path between end parties.

The IP media path is established directly between the originating and terminating parties on the end nodes. No media resources are used on the tandem switch. When trunks are not optimized, signaling continues to be handled in a tandem manner, even though the media path is direct.

Tandem operations

All media paths route directly over IP networks. However, to maintain proper control points and billing records for a call, sometimes signaling must be indirect. The following sections describe indirect signaling operations for these scenarios.

Direct tandem calls Because IP Peer Networking uses an NRS (H.323 Gatekeeper) for address resolution, the requirement for tandem calls is minimal. With an NRS (H.323 Gatekeeper), each node can obtain the IP address of the terminating node. Therefore, calls route directly to the terminating node and not through a tandem node.

Feature modification (for example, Call Transfer) can cause calls to tandem. Tandem calls also occur when routing is configured as tandem, so accounting records can generate during calls from a third-party gateway.

Tandem feature calls When a tandem call occurs due to a transfer operation, the IP media paths between the originating party and the "transferred-to" party must be redirected to each other. This redirection is initiated by the transferring (tandem) node.

This scenario describes a form of Trunk Route Optimization (TRO).

When a tandem call occurs due to a Call Forward operation, it attempts to use TRO to optimize the route between the originating and "transferred-to" parties. If packaging or user provisioning selections mean that TRO is not supported, the tandem node initiates media path redirection for both parties.

TRO is used when a call from Node A to Node B forwards to Node C. Node B sends a TRO facility message to Node A. The message contains the digits of the "forwarded-to" party. Node A resolves these digits to a route and determines whether it has a direct route configured to Node C.

IP Peer handling of TRO differs slightly from the PRI handling at this point. Unlike the Primary Rate case where each destination has a dedicated route and ISDN link, for IP Peer, in Node A’s routing configuration, all remote locations are reached using the same Virtual Trunk (the H.323 Gatekeeper subsequently translates the digits to separate IP nodes). When TRO is attempted at Node A, the call processing finds that the new destination is accessed through the same Virtual Trunk route, and accepts the TRO even though the call does not have an alternate direct route to Node C. The tandem call routing through Node B is cleared. Node A places a new call through the same Virtual Trunk route and IP D-channel that was used for the original call to Node B. H.323 Gatekeeper translation identifies the correct destination, Node C, and the call is placed directly to that node.

In cases where the TRO feature does not optimize trunks, the Virtual Trunks must remain busy at Nodes A, B and C until the call is released. A direct media path between Node A and Node C supports the connection; Node B is not on the media path. This eliminates voice quality problems caused by multiple transcoding steps.

TRO versus TAT on transferred call The TRO feature will optimize a redirected call initiated on the transfer key. Station A calls Station B on Node 1. Station B puts the call on hold and initiates a call transfer over an H.323 trunk to Station C on Node 2, which call forwards no answer to Station D on Node 1. While Station D is ringing (Station B hasn’t completed the call transfer), optimization will be done by the TRO feature. If Station B completes the call transfer while Station C is ringing, and then the call is forwarded no answer to Station D, the TRO feature will optimize the redirected call and release the two H.323 trunks connecting Node 1 and Node 2 before Station D answers the call.

Circuit-switched tandem calls The IP Peer Networking feature supports circuit-switched tandem calls by configuring a circuit-switched TIE trunk on a CS 1000 system, or gateway which routes calls across the IP network. The signaling over the circuit-switched trunk can use any of the TIE trunks supported in traditional MCDN circuit-switched networks.

Virtual Trunk calls in conference

A party on Node A calls a party on Node B. The party on Node B creates a three-party conference with a party on Node C. A circuit-switched conference circuit is used on Node B. Each party has their media path redirected to a separate circuit-switched gateway on Node B. Circuit-switched speech paths are established between each circuit-switched gateway and the conference bridge.

Virtual Trunk to circuit-switched party transferred to an IP Phone

A call is established between a party on a remote node and a circuit-switched party on the local node using a Virtual Trunk. A media path exists between the remote party (the remote party can be an IP Phone or a circuit-switched gateway) and a circuit-switched gateway on the local node.

The local circuit-switched party transfers the call to an IP Phone on the local node. When the circuit-switched party initiates a transfer operation, call processing on the local node places the remote party on hold, according to existing functionality. H.323 signaling places the remote party in a "paused" state, and the existing media path remains allocated. A local call is set up between the transferring circuit-switched party and the local IP Phone.

When the circuit-switched party completes the transfer, the consultation call is released, and a call is set up between the remote party and the transferred-to party. The media path (that existed between the remote party and the transferring circuit-switched party) is redirected using the H.323 pause and re-routing mechanism. As the transferred-to party is not a circuit-switched telephone, the circuit-switched gateway resource is released. The call scenario completes with a direct media path between the remote party and the IP Phone on the local node.

Virtual Trunk to a circuit-switched party transferred before answer to an IP Phone

A call is established between a party on a remote node and a circuit-switched party on the local node over a Virtual Trunk. A direct IP media path exists between the remote party (for example, an IP Phone or circuit-switched gateway) and a circuit-switched gateway on the local node. The local circuit-switched party initiates a transfer to an IP Phone on the local node. While the IP Phone is ringing, the transferring party completes the transfer by disconnecting or pressing the Transfer key. The originating party receives ringback tone.

When the circuit-switched party initiates the Transfer operation, the incoming Virtual Trunk (and indirectly, the originating party) is placed on hold and the direct IP media path between the originating party and the circuit-switched gateway is torn down. If Music or RAN is configured, a new IP media path is established between a circuit-switched gateway and the originating party.

When the transferring party completes the "transfer before answer", ringback tone must be provided to the originating party. A new IP media path is established between a circuit-switched gateway on the node providing the ringback tone and the originating party. The media path is one way only, from the circuit-switched gateway to the originating party. The node providing the ringback tone initiates this media path "redirection" using the H.323 "Third-party initiated pause and re-routing" mechanism. It does not use ISDN Q.931 signaling for this purpose.

When the party on the IP Phone answers, another media path redirection occurs. The media path between the circuit-switched gateway and the originating party is released, and a new two-way IP media path is established between the originating party and the IP Phone party. This uses the H.323 "Third-party initiated pause and re-routing" mechanism.

IP Phone to local IP Phone transferred to a Virtual Trunk

A call is established between two IP Phones on the same node. A direct media path exists between the two telephones. One of the parties initiates a transfer to a party on a remote node.

When the IP Phone party initiates the transfer, call processing on the local node places the other party on hold. The media path between the two IP Phones is torn down. A call is set up between the transferring IP Phone and the remote party (this could be an IP Phone or circuit-switched telephone). See "IP Phone to IP Phone (on separate Call Servers)" (page 46).

When the transferring IP Phone completes the transfer before answer, the consultation call between the IP Phone and the remote party is torn down and a call is set up between the transferred IP Phone and the remote party. The media path that existed between the remote party and the transferring IP Phone is redirected using the H.323 third-party initiated pause and re-routing mechanism. No ISDN Q.931 signaling is exchanged between the nodes, and the call state on the terminating node is unchanged. A direct IP media path is established between the transferred IP Phone and the remote party.

H.323-to-SIP signaling

Contents

This section contains information on the following topics:

Introduction

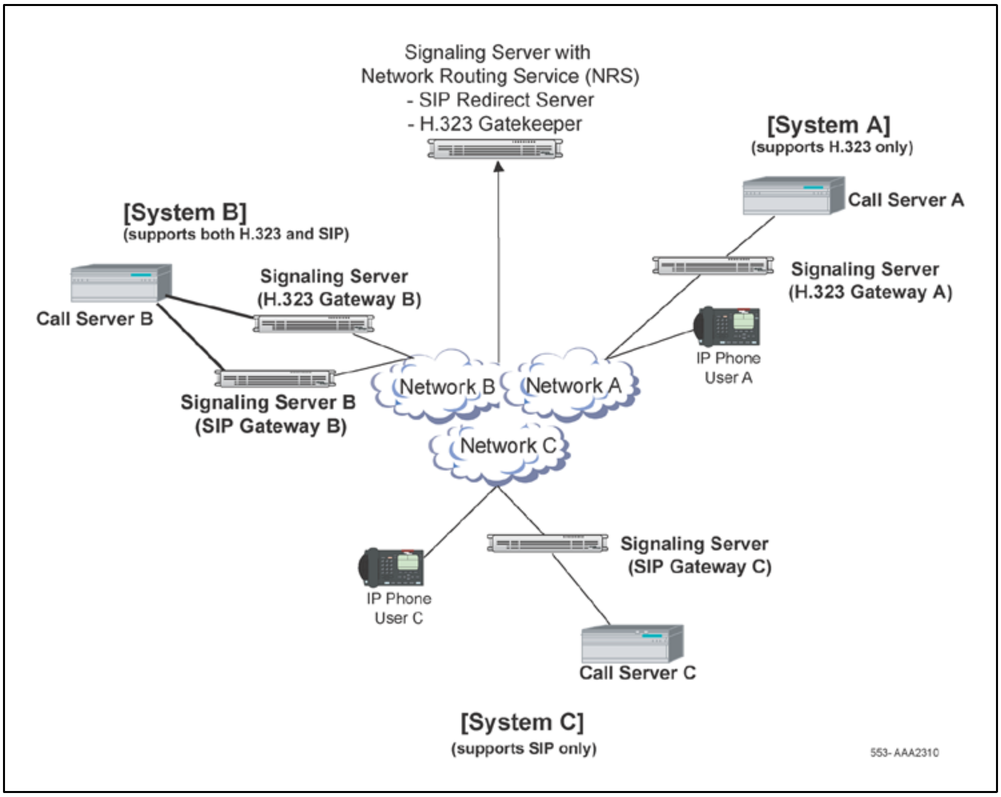
H.323 is used to set up and tear down H.323 calls, while SIP is used to set up and tear down SIP calls. If a network uses both the SIP and H.323 protocols, then an H.323-to-SIP "bridge" must exist between the H.323 domain and the SIP domain.

H.323-to-SIP signaling (coexistence of both H.323 and SIP)

This section describes a call flow example of an H.323 incoming trunk call to a SIP trunk. In the following example, System A supports only H.323, System B supports both SIP and H.323, and System C supports only SIP. See Figure 25 "Sample network for H.323-to-SIP call" (page 82).

> ***Note:*** In CS 1000 Release 5.0, RFC2833 provides digit handling for DTMF signaling. With RFC2833, CS 1000 systems can inter-operate with SIP-based devices that do not support out-of-band DTMF digit signaling. RFC2833 can be used for SIP calls only. H.323 and local calls are not supported.

Figure 25
Sample network for H.323-to-SIP call



In this example, System B shows two Signaling Servers:

- one serves the H.323 Virtual Trunk
- another serves the SIP Virtual Trunk

The Signaling Servers in System B are shown as two separate servers for clarity. Both the H.323 Gateway and the SIP Trunk Gateway can be configured on the same Signaling Server. Each Signaling Server has its own D-channel IP, and both are connected to the same Call Server.

> ***Note:*** This statement does not imply that H.323 and SIP cannot coexist on one Signaling Server. If both applications are enabled, then the two Signaling Servers in Figure 25 "Sample network for H.323-to-SIP call" (page 82) will collapse into one Signaling Server.

In this example, System C (which is the SIP domain) is a CS 1000 system. However, System C could be any type of SIP endpoint such as a SIP Phone or MCS 5100 system.

The implementation of H.323-to-SIP basic call flow is similar to an H.323 Virtual Trunk-to-Virtual Trunk tandem call (see "Virtual Trunk to Virtual Trunk" (page 58) for SIP and "Virtual Trunk to Virtual Trunk" (page 77) for H.323).

The difference is in the SIP Network Protocol Module (NPM) (that is, the SIP Trunk Gateway, where the ISDN messages are converted to the corresponding SIP messages).

Call scenarios summary

Using the configuration shown in Figure 25 "Sample network for H.323-to-SIP call" (page 82), the following call scenarios exist:

- Calls between System A (H.323) and System C (SIP) are not possible, because each system supports a different protocol.
- H.323 calls between System A and System B are possible. SIP calls between System A and System B are not possible, because System A does not support SIP.
- SIP calls between System B and System C are possible. H.323 calls between System B and System C are not possible, because System C does not support H.323.
- Call between System A and System C are possible when routed through System B, because the Call Server in System B can convert H.323 calls to SIP and SIP calls to H.323. Therefore, a SIP call from System C is converted to H.323 in System B and terminates at System A. Similarly, an H.323 call from System A is converted to SIP in System B and terminates at System C. This scenario is a genuine SIP/H.323 network.

Call walk-through

IP Phone A (which has H.323-only configuration) wants to talk to IP Phone C (which has SIP-only configuration).

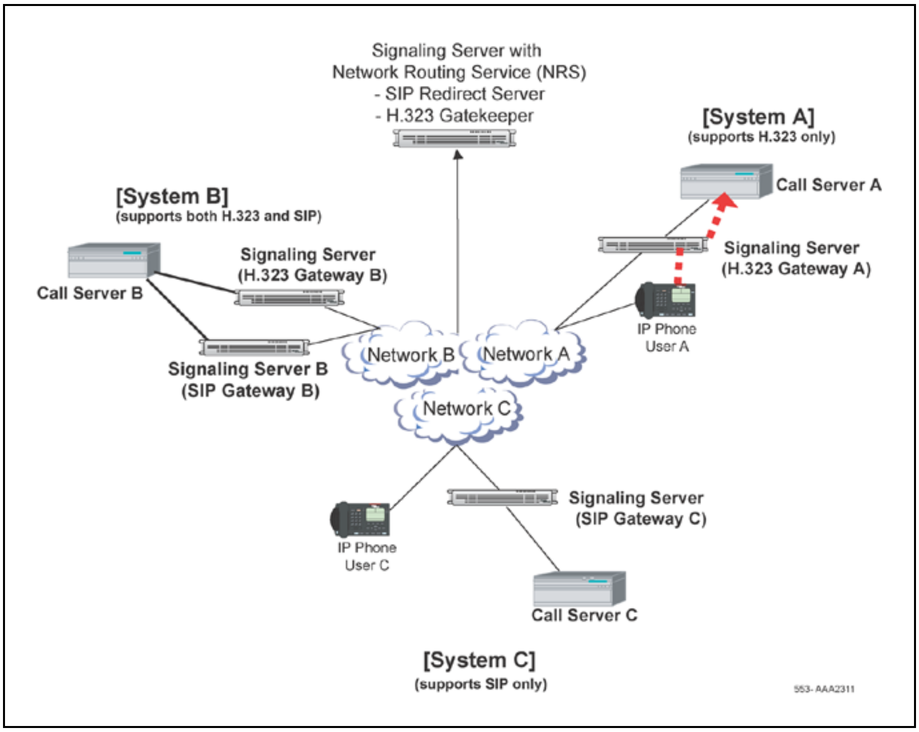
The following scenario describes the Direct IP Media Path functionality for a basic network call.

> ***Note:*** Only the primary messages are illustrated in the following call flows.

1. User A on Call Server A dials the DN of User C on Call Server C. In order to get to User C, the call must go through System B for digit manipulation. See Figure 26 "User A dials User C" (page 84).

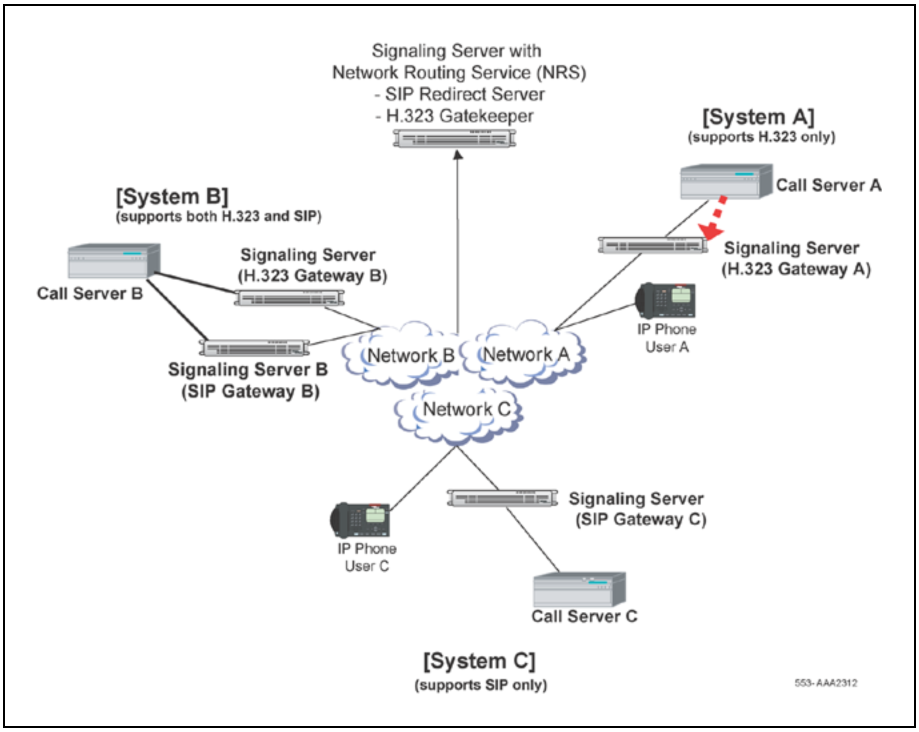
   > ***Note:*** The following call walk-through assumes that System A is using an H.323 Gateway only, System C is using a SIP Trunk Gateway only, and System B has both an H.323 Gateway and a SIP Trunk Gateway.

Figure 26
User A dials User C



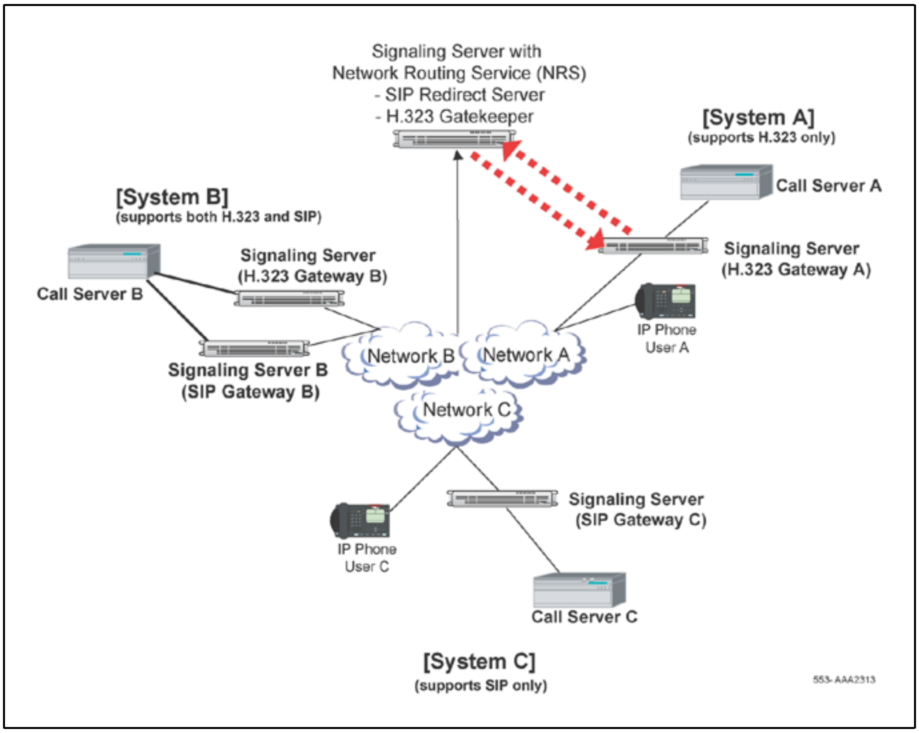
2. Call Server A determines that the dialed digits are at another site. Call Server A select the codec list, allocates bandwidth, and routes the call to the IP network using a Virtual Trunk and H.323 Gateway A. See Figure 27 "Call Server A routes the call to the IP network" (page 85).

Figure 27
Call Server A routes the call to the IP network



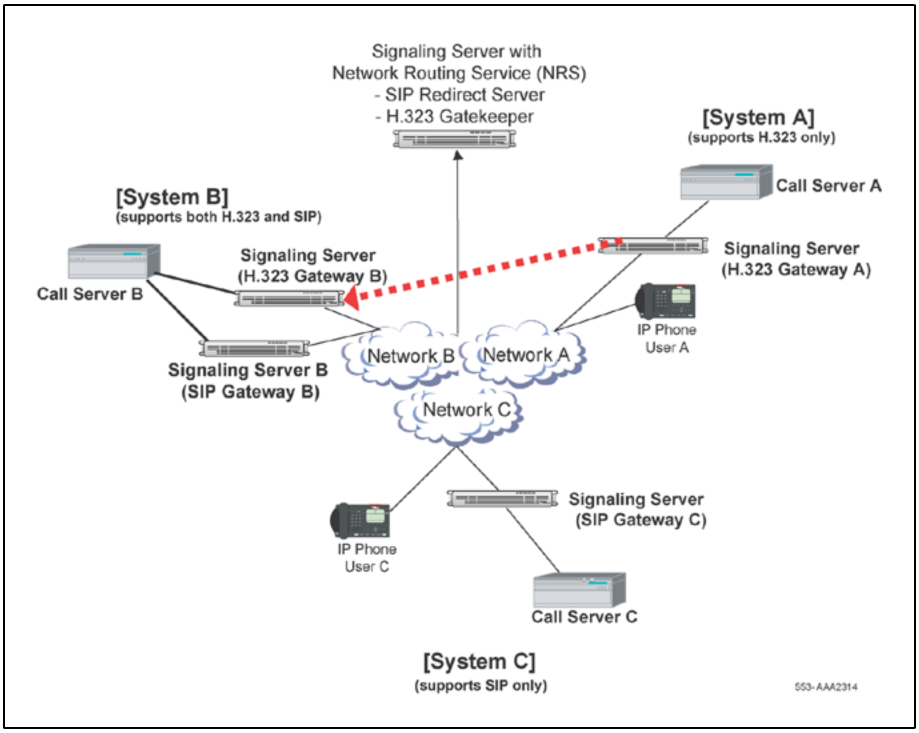
3. H.323 Gateway A asks the NRS (H.323 Gatekeeper) to search for the dialed DN in its database, as System A cannot go directly to System C because System A is using H.323 only and System C is using SIP. The NRS (H.323 Gatekeeper) responds back to H.323 Gateway A with the IP address of the H.323 Gateway B in System B. See Figure 28 "H.323 Gateway A communicates with the NRS (H.323 Gatekeeper)" (page 86).

Figure 28
H.323 Gateway A communicates with the NRS (H.323 Gatekeeper)



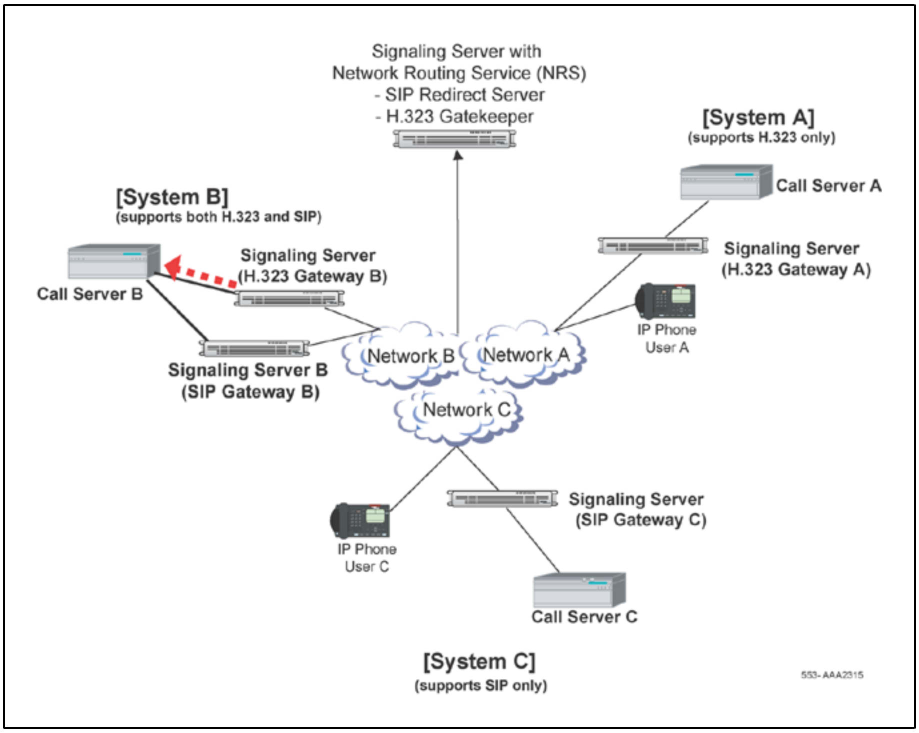
4. H.323 Gateway A sends an H.323 SETUP message to H.323 Gateway B including the DN information and IP Phone information (IP address and port number) for User A. See Figure 29 "H.323 Gateway A sends information to H.323 Gateway B" (page 87).

Figure 29
H.323 Gateway A sends information to H.323 Gateway B



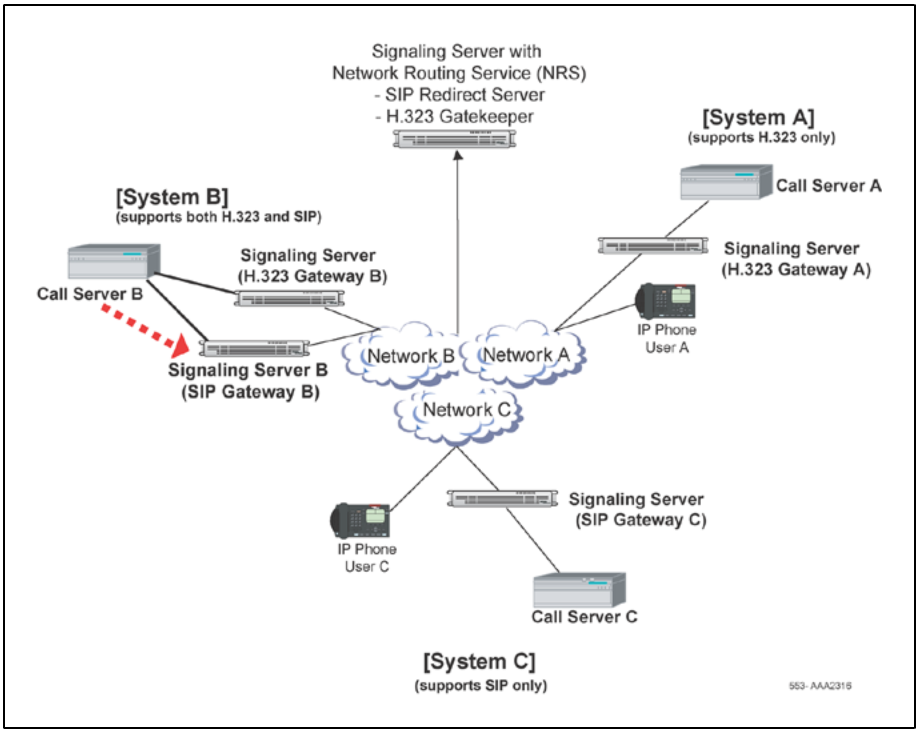
5. H.323 Gateway B receives the message from H.323 Gateway A and sends the call to Call Server B over a Virtual Trunk. Call Server B also treats the call as an incoming call from a Virtual Trunk. See Figure 30 "H.323 Gateway B sends calls to Call Server B" (page 88).

Figure 30
H.323 Gateway B sends calls to Call Server B



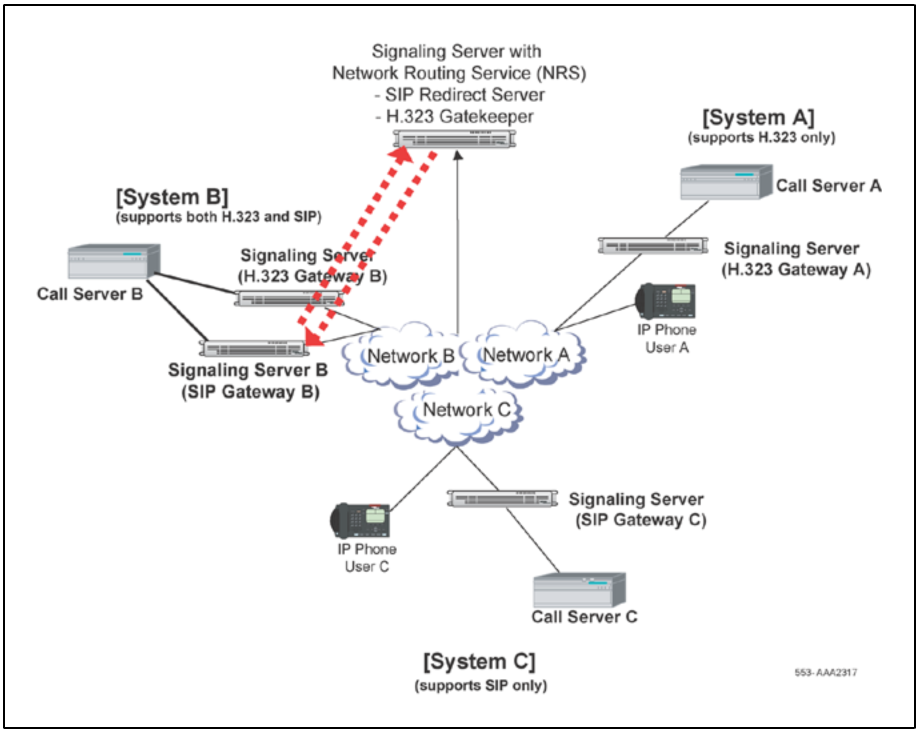
6. Call Server B processes the incoming message and determines that the call should go to System C through SIP Trunk Gateway B. Call Server B routes the call to SIP Trunk Gateway B. See Figure 31 "Call Server B sends calls to SIP Trunk Gateway B" (page 89).

Figure 31
Call Server B sends calls to SIP Trunk Gateway B



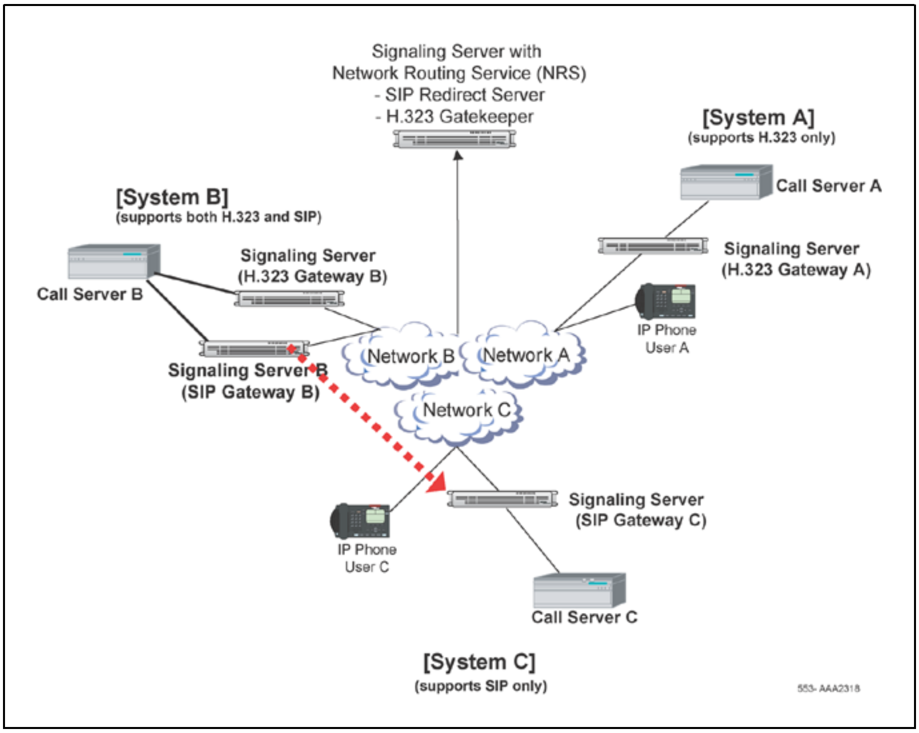
7. SIP Trunk Gateway B asks the NRS (SIP Redirect Server) to do a search for the DN of User C. The NRS (SIP Redirect Server) sends the IP address of SIP Trunk Gateway C to SIP Trunk Gateway B. See Figure 32 "SIP Trunk Gateway B communicates with the NRS (SIP Redirect Server)" (page 90).

Figure 32
SIP Trunk Gateway B communicates with the NRS (SIP Redirect Server)



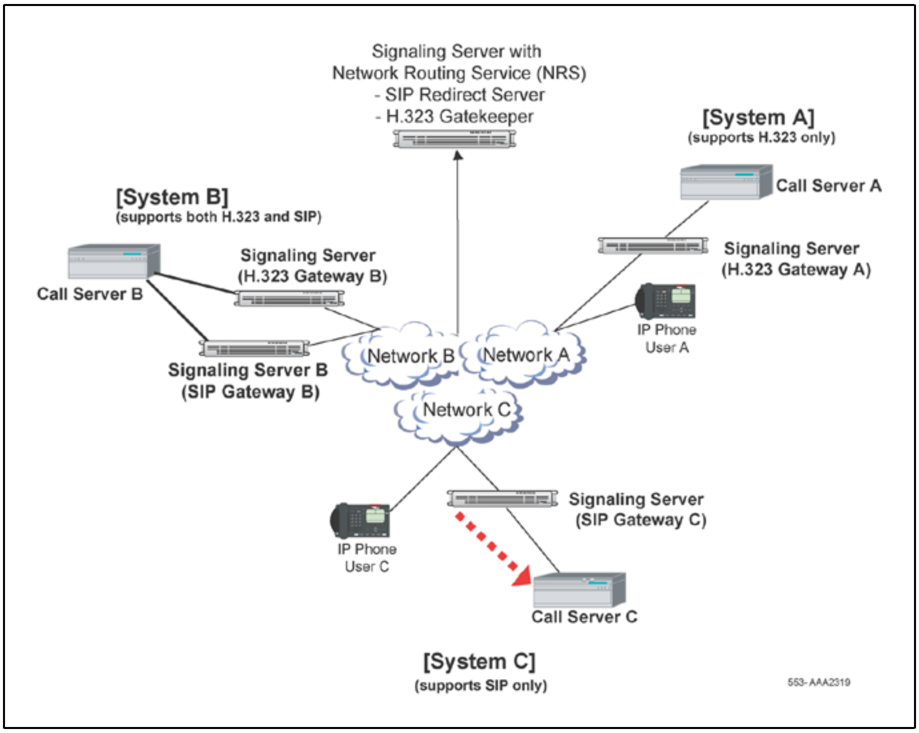
8. SIP Trunk Gateway B sends an INVITE message to SIP Trunk Gateway C. See Figure 33 "SIP Trunk Gateway B sends INVITE message to SIP Trunk Gateway C" (page 91).

Figure 33
SIP Trunk Gateway B sends INVITE message to SIP Trunk Gateway C



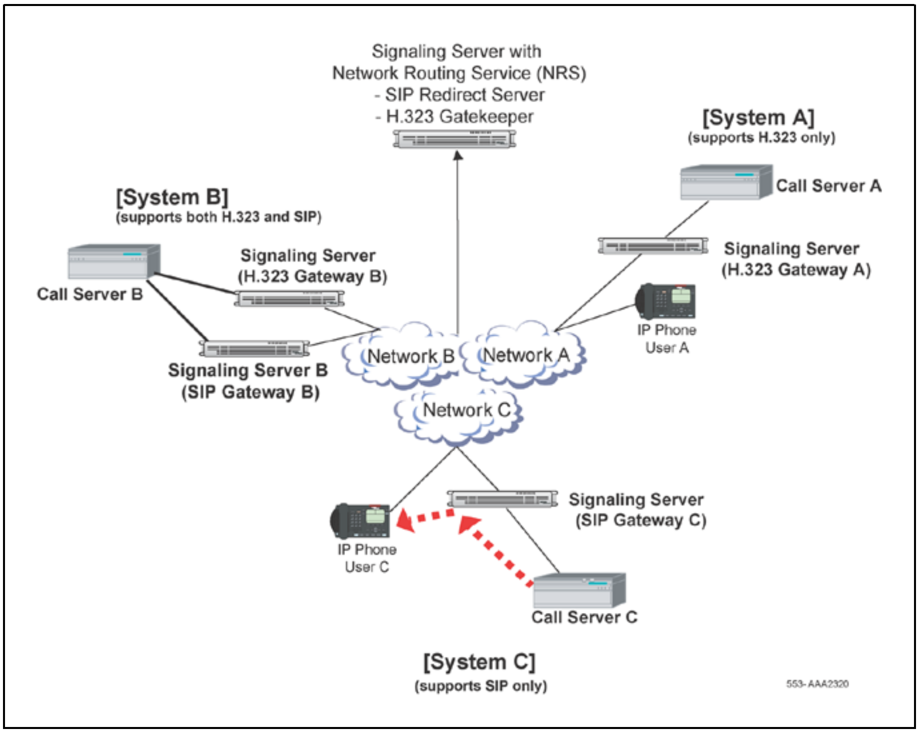
9. SIP Trunk Gateway C sends the call to Call Server C. See Figure 34 "SIP Trunk Gateway C sends call to Call Server C" (page 92).

Figure 34
SIP Trunk Gateway C sends call to Call Server C



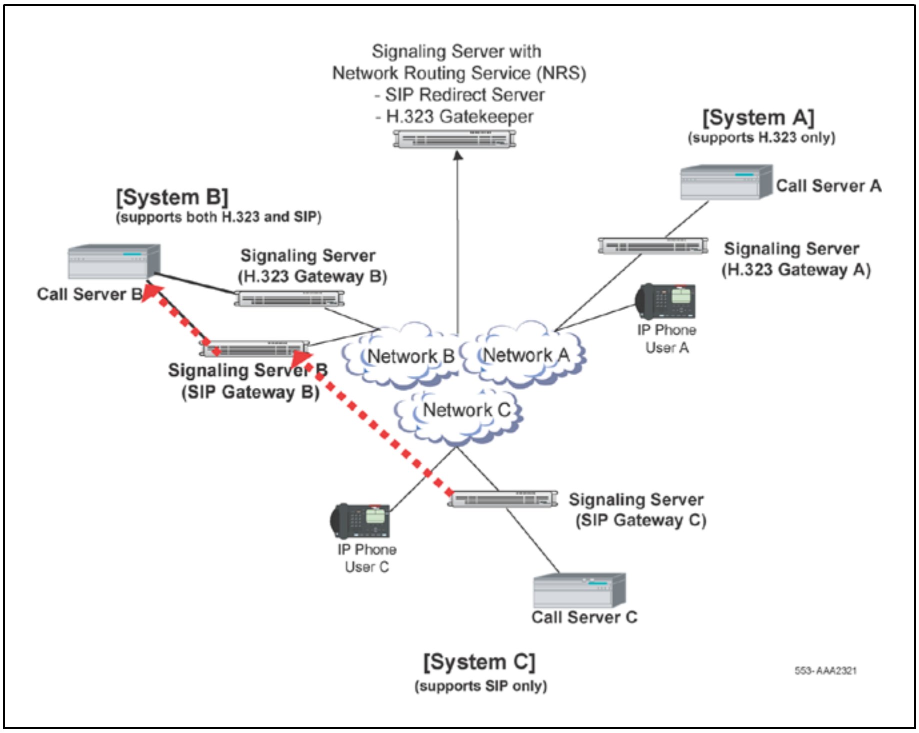
10. Call Server C selects the codec, allocates bandwidth, rings the telephone, and sends an ISDN Alert message to SIP Trunk Gateway C. See Figure 35 "Call Server C sends Alert message to SIP Trunk Gateway C" (page 93).

Figure 35
Call Server C sends Alert message to SIP Trunk Gateway C



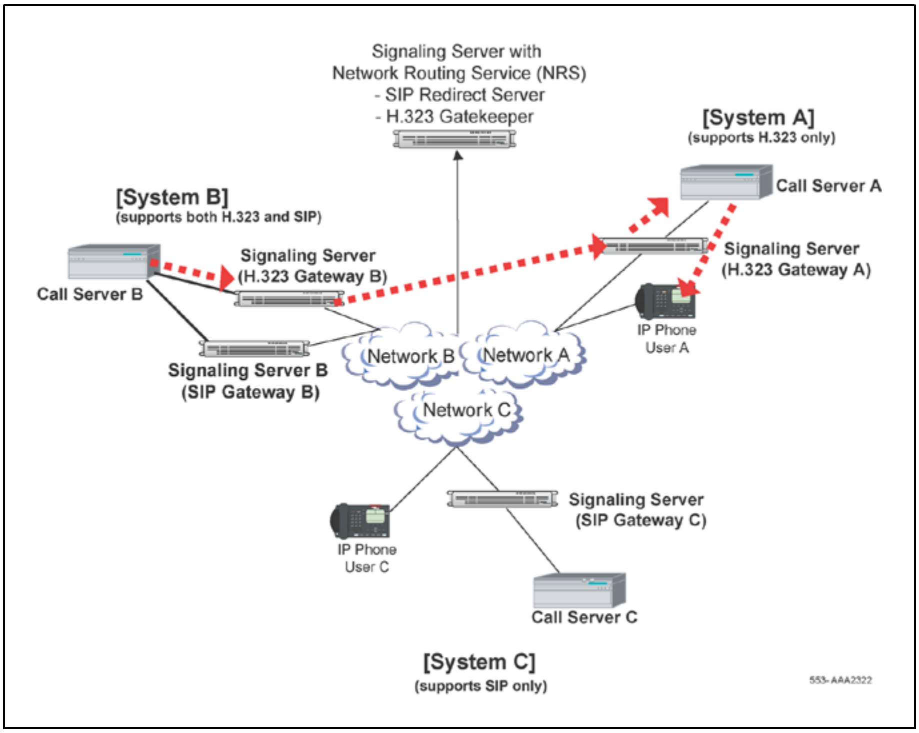
11. SIP Trunk Gateway C converts the ISDN Alert message to a SIP 180 response message. SIP Trunk Gateway C sends the SIP message to SIP Trunk Gateway B. SIP Trunk Gateway B converts the incoming SIP 180 response message back to the ISDN Alert message. SIP Trunk Gateway B then sends the message to Call Server B. See Figure 36 "SIP Trunk Gateway B sends ISDN Alert message to Call Server B" (page 94).

Figure 36
SIP Trunk Gateway B sends ISDN Alert message to Call Server B



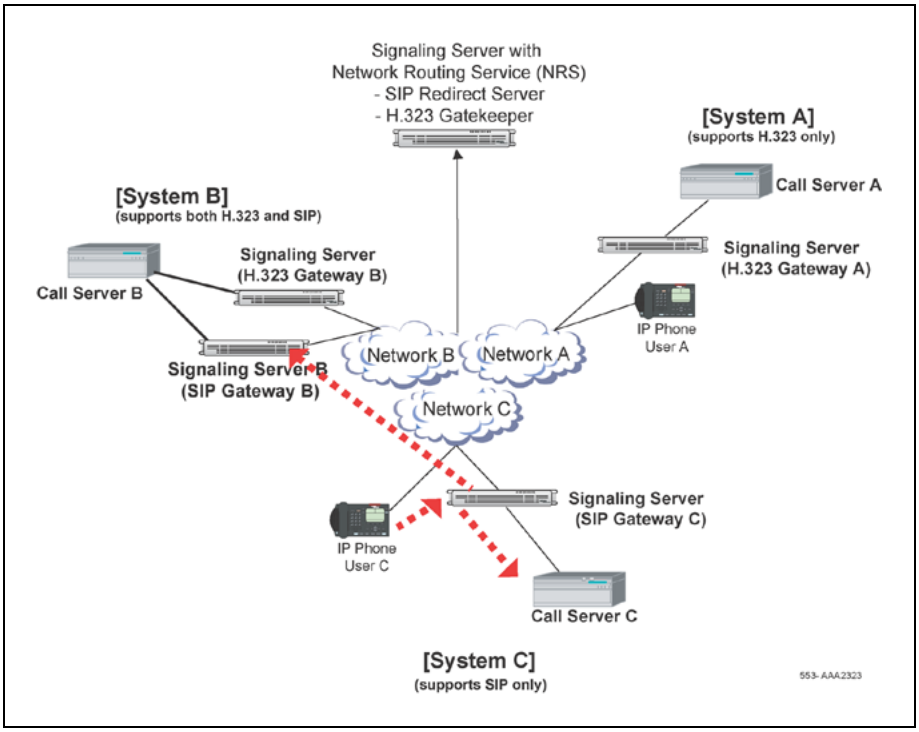
12. Call Server B forwards the ISDN Alert message to H.323 Gateway B. H.323 Gateway B sends the message to H.323 Gateway A. H.323 Gateway A sends the message to Call Server A. Call Server A requests that IP Phone User A play ringback tone. See Figure 37 "H.323 Gateway B sends Alert message to H.323 Gateway A" (page 95).

Figure 37
H.323 Gateway B sends Alert message to H.323 Gateway A



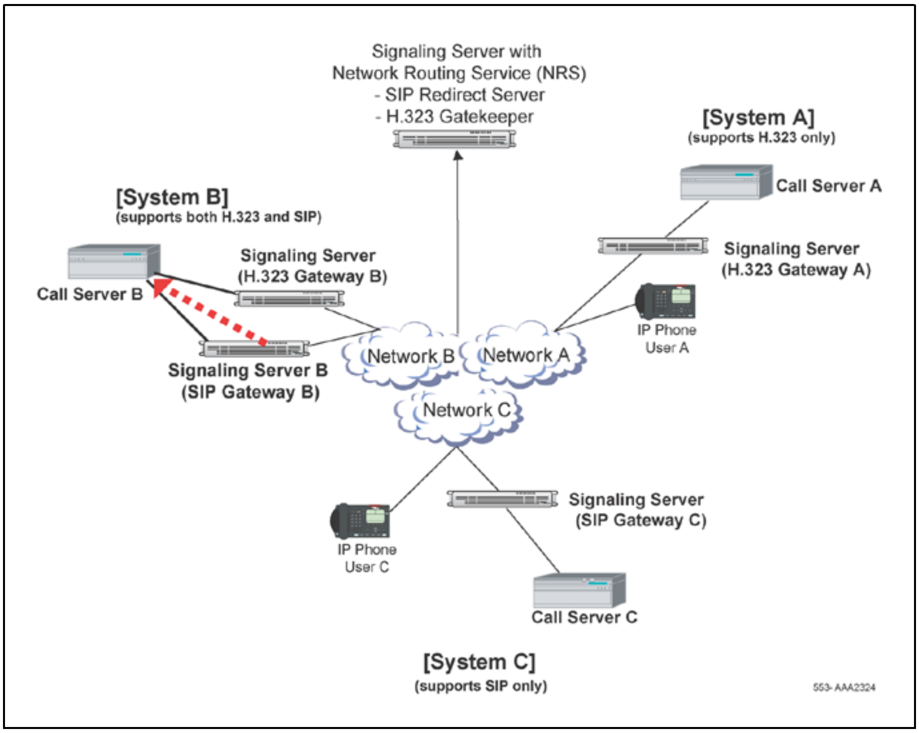
13. IP Phone User C answers the call. A message is sent to Call Server C on SIP Trunk Gateway C. SIP Trunk Gateway C sends a SIP 200 OK message along with the IP Phone information (IP address, port numbers, and codec) to SIP Trunk Gateway B. See Figure 38 "User C answers the call" (page 96).

Figure 38
User C answers the call



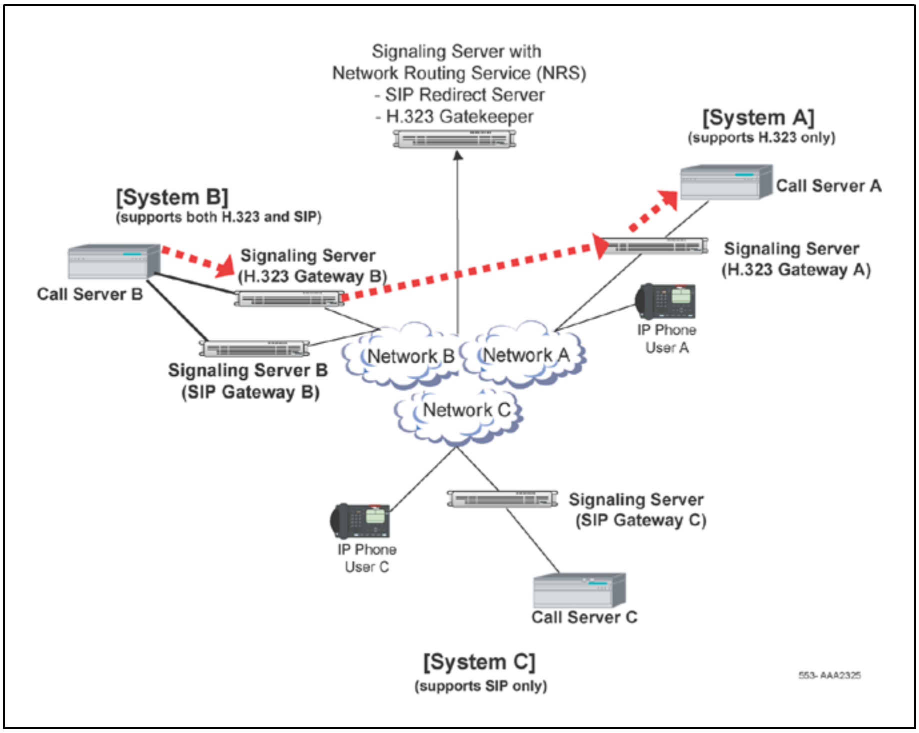
14. SIP Trunk Gateway B converts the SIP 200 OK message to an ISDN CONNECT message and sends the message to Call Server B. See Figure 39 "SIP Trunk Gateway B sends message to Call Server B" (page 97).

Figure 39
SIP Trunk Gateway B sends message to Call Server B



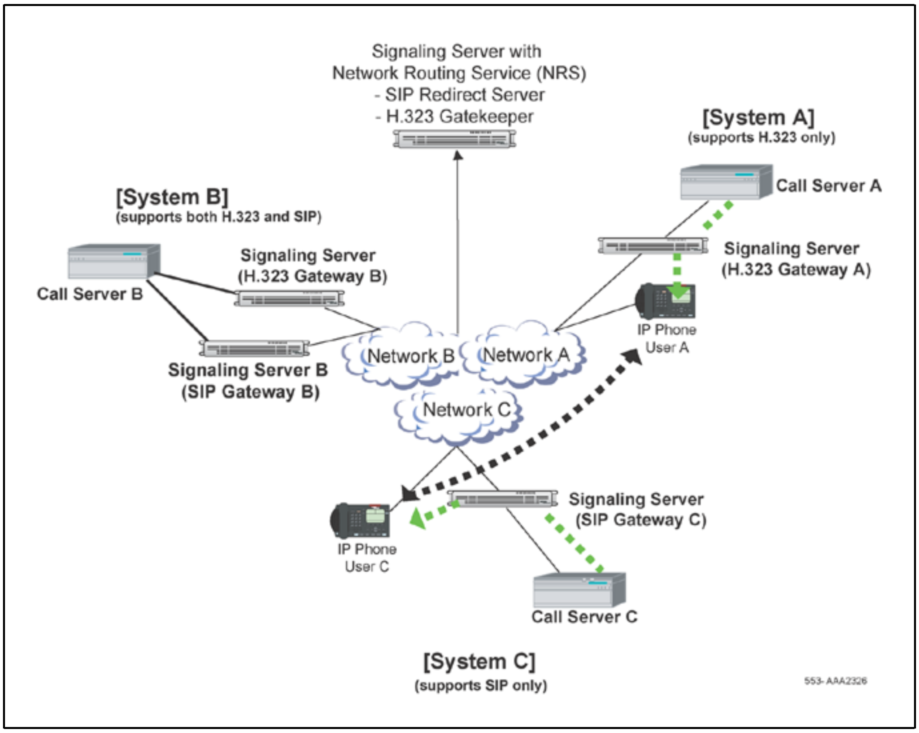
15. Call Server B forwards the ISDN CONNECT message to H.323 Gateway B. H.323 Gateway B then sends a message to H.323 Gateway A. H.323 Gateway A sends the message to Call Server A. See Figure 40 "H.323 Gateway B sends message to H.323 Gateway A" (page 98).

Figure 40
H.323 Gateway B sends message to H.323 Gateway A



16. Call Server A tells IP Phone A to set up the direct IP media path with IP Phone C. The IP Phones then begin to transmit and receive voice over the IP network. See Figure 41 "IP Phones start the direct media paths" (page 99).

Figure 41
IP Phones start the direct media paths



Fallback to PSTN

Contents

This section contains information on the following topics:

Introduction

It is possible to automatically Fallback to PSTN, if calls cannot be completed due to loss of connectivity between sites over the IP network. This is achieved using the standard MCDN Alternate Routing feature when:

- the IP network is down
- the destination IP Peer endpoint is not responding
- the destination IP Peer endpoint responds that there are no available IP Peer trunk resources
- the destination IP Peer endpoint is not registered with the NRS
- there are address translation errors
- all Virtual Trunks are busy at the originating sites
- all bandwidth configured for a bandwidth zone has been allocated
- Quality of Service (QoS) metrics cause a reduction in available bandwidth (see "Fallback to PSTN" (page 101))

Fallback to PSTN can be configured by programming an alternate route entry after the virtual IP trunk route entry in RLB in LD 86 and entering RRA at the SBOC prompt for the virtual IP trunk entry. Refer to *Software Input/Output: Administration* (NN43001-611) for the configuration of RLB in LD 86.

Fallback to PSTN for IP Peer Networking refers to the use of the MCDN Alternate Routing feature to step back to any alternate switched-circuit trunk route to the destination that the call first attempted to reach by the IP Peer virtual IP trunk route.

The alternate switched-circuit trunk route can be any of the following:

- a direct ISDN PRI tie trunk route
- a Virtual Private Voice Network tie trunk route using a common carrier voice network
- a PSTN trunk route

> ***Note 1:*** If Fallback to PSTN uses PSTN trunks as the alternate route, then the appropriate ESN digit manipulation features must be implemented to convert the dialed number from on-net to off-net, or from private to public E.164 format.
>
> ***Note 2:*** If Fallback to PSTN uses PSTN trunks as the alternate route, Nortel recommends that you configure both the original and alternate trunk routes as en bloc-capable or overlap-capable. See "Overlap signaling" (page 251) for more information.

A similar feature, Alternative Call Routing for Network Bandwidth Management, is available to provide alternate routing between a branch office (or Survivable Remote Gateway [SRG]) and a main office. Refer to *Branch Office Installation and Commissioning* (NN43001-314) for more information.

Engineering practices

Best IP network engineering practices for IP Telephony

In general, the best IP network engineering practices for IP Telephony tend to remove the requirement for QoS Fallback to PSTN. Best practices include:

- implementing network QoS features such as DiffServ and 802.1Q to give priority to real-time voice traffic
- fragmenting large data frames to limit the maximum frame size on low speed WAN links and limiting the quantity of voice traffic that is transmitted over low speed links

When QoS Fallback to PSTN is required for certain network locations (in an IP Peer network) because WAN links have not been engineered according to best practices, IP Trunk 3.0 (or later) can be used to achieve QoS Fallback to PSTN between those locations and an IP Peer node located on the IP network backbone. An IP Trunk 3.0 (or later) node must be configured in the same CS 1000 system with the IP Peer node.

Engineering considerations for using IP Trunk to achieve QoS Fallback to PSTN

Using IP Trunk 3.0 (or later) nodes to provide QoS Fallback to PSTN in an IP Peer network imposes certain engineering and network management trade-offs that must be carefully considered:

- QoS Fallback to PSTN only works between symmetrically-configured pairs of IP Trunk nodes. QoS Fallback to PSTN does not work between an IP Trunk node and an IP Peer node. Each IP Trunk node in a symmetrically-configured pair must have QoS Fallback to PSTN enabled for the opposite destination node.
- A pair of symmetrically-configured IP Trunk nodes must each have a local Dialing Plan entry in the IP Trunk node that points to these opposite IP Trunk nodes. The Gatekeeper cannot be used for any IP Trunk destinations that are symmetrically configured to enable QoS Fallback to PSTN.
- An IP Trunk node configured in a CS 1000 system with an IP Peer node does not support the Direct Media Path feature of IP Peer Networking. Therefore all IP Trunk calls originating or terminating at the network location that require QoS Fallback to PSTN must have a tandem media path connection through the CS 1000 IP Peer node. The tandem media path can occasionally cause voice quality degradation due to multiple transcoding and higher end-to-end latency of the voice conversation.

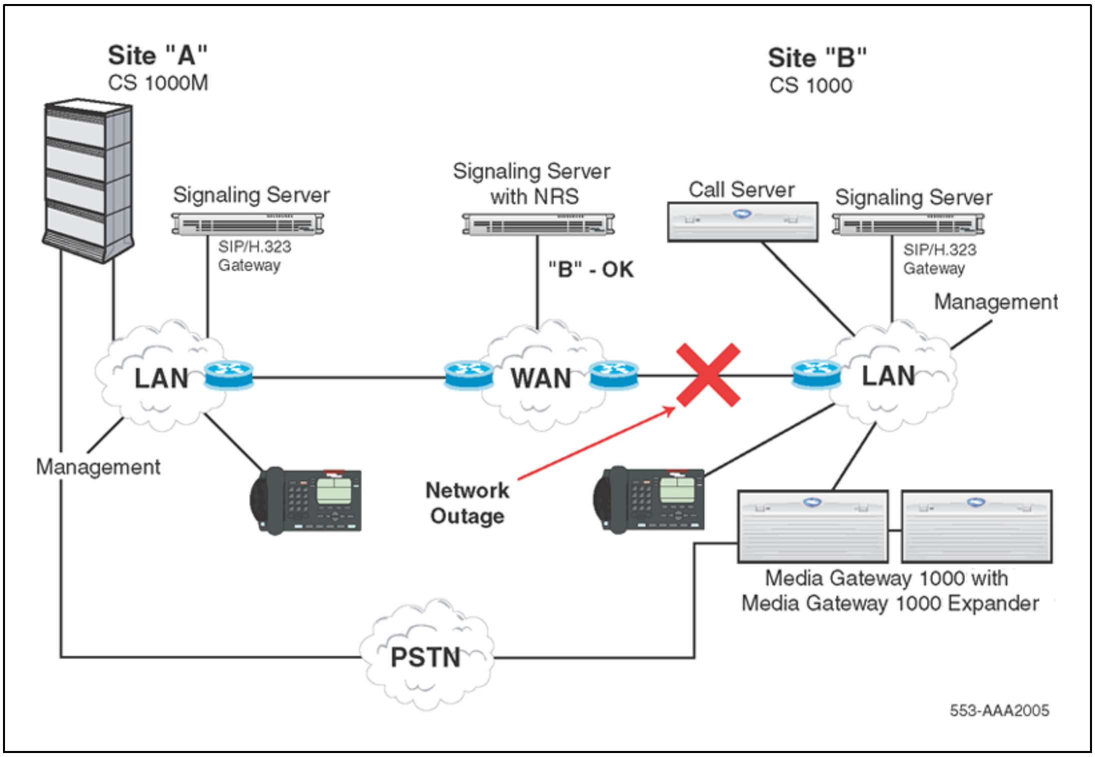
For more information, refer to *IP Trunk: Description, Installation, and Operation* (NN43001-563) and *Basic Network Features* (NN43001-579).

Alternate circuit-switched routing

The following scenario describes alternate circuit-switched routing when there is an IP network outage:

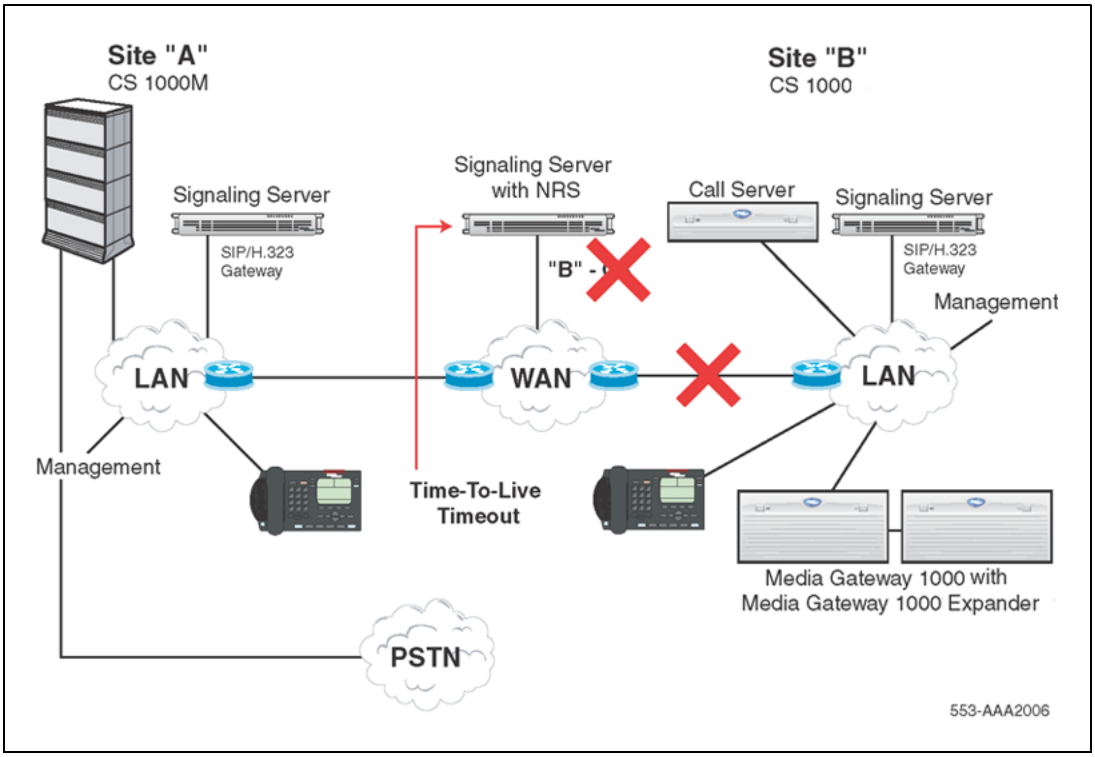
1. An IP network outage occurs at Site B. See Figure 42 "IP network outage at Site B" (page 104).

Figure 42
IP network outage at Site B



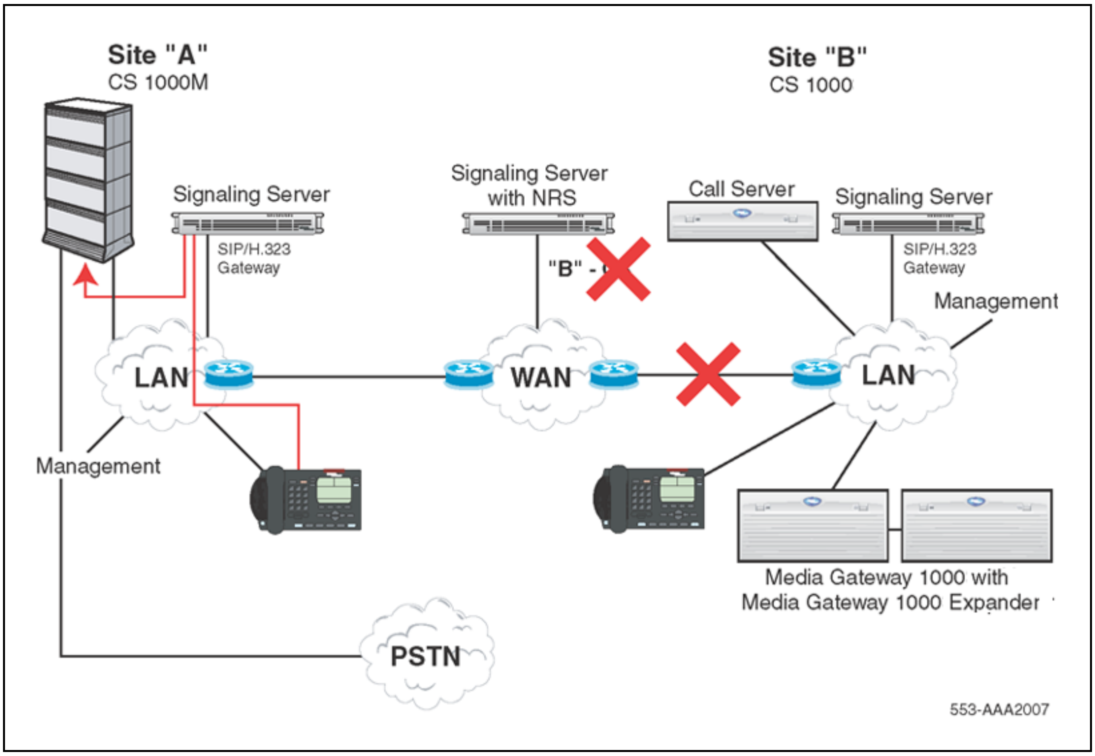
2. The registration of Site B times out at the NRS; the status updates. See Figure 43 "Registration at Site B times out" (page 105).

Figure 43
Registration at Site B times out



3. User A on Call Server A dials the DN of User B on Call Server B. Call Server A collects the dialed digits through the Terminal Proxy Server (TPS) on the Signaling Server. See Figure 44 "User A dials User B" (page 106).

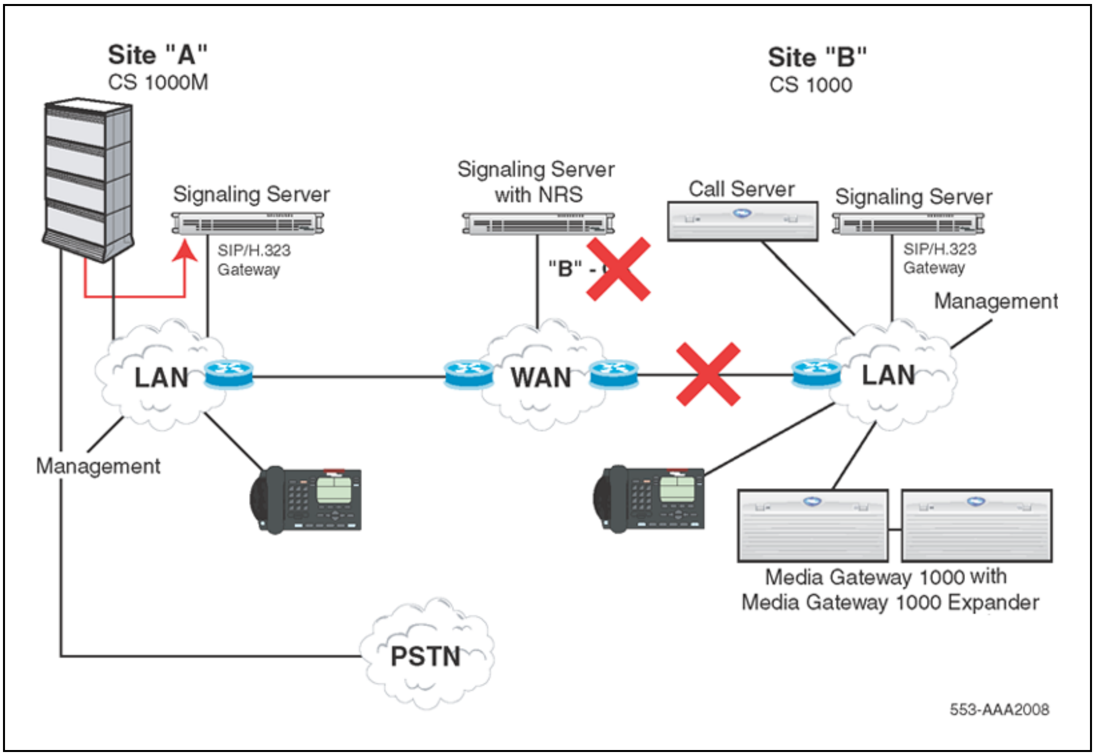
Figure 44
User A dials User B



4. Call Server A determines that the DN is at another site. Call Server A selects the codec list, allocates bandwidth, and routes the call to the IP network, using a Virtual Trunk and the SIP/H.323 Gateway. See Figure 45 "Call Server A routes the call to the IP network" (page 107).

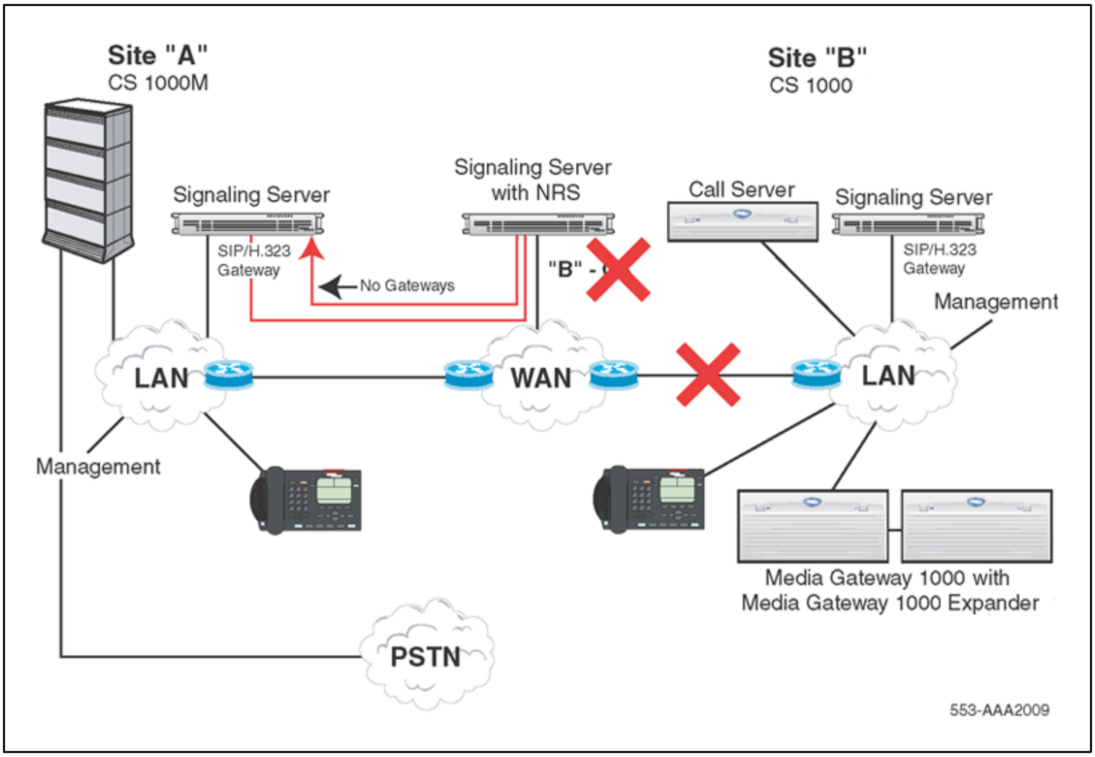
   Note: To select which Virtual Trunk to use for routing, Call Server A examines the number dialed and uses various trunk routing and signaling features (for example, ESN and MCDN).

Figure 45
Call Server A routes the call to the IP network



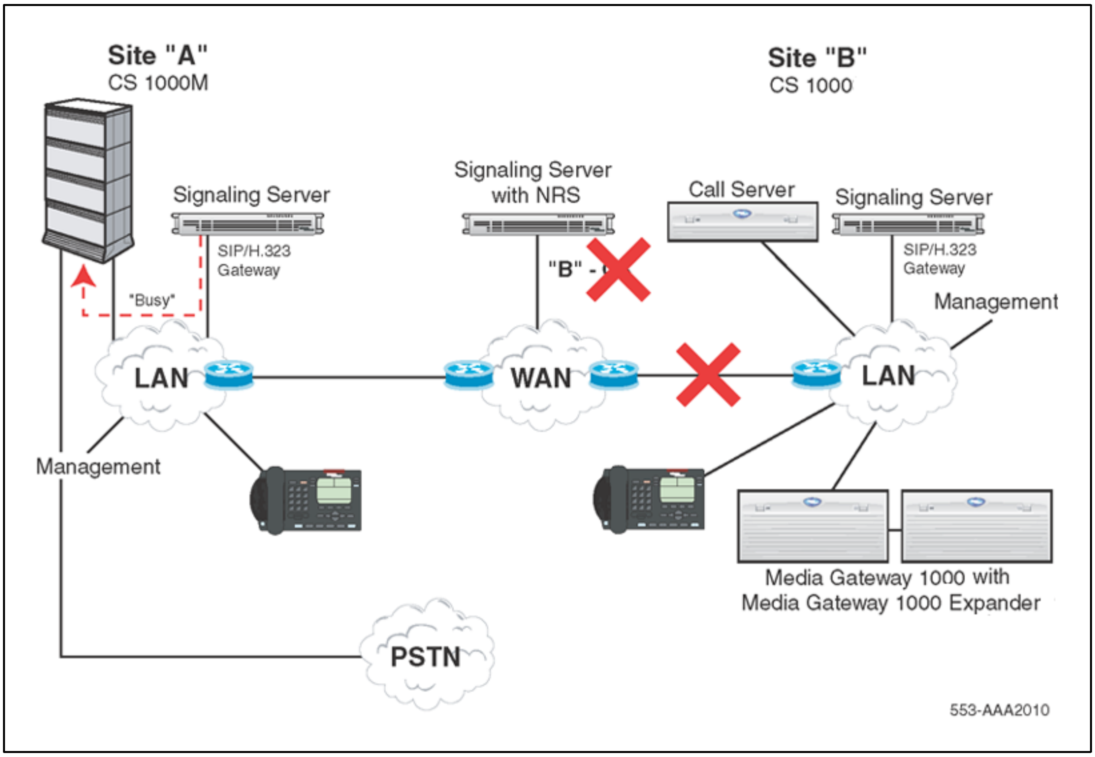
5. SIP/H.323 Gateway A asks the NRS to search for a dialed DN in the database (for example, within the appropriate CDP domain). The NRS replies that no SIP/H.323 Gateways are available for the dialed number. See Figure 46 "No SIP/H.323 Gateways are available for the dialed DN" (page 108).

Figure 46
No SIP/H.323 Gateways are available for the dialed DN



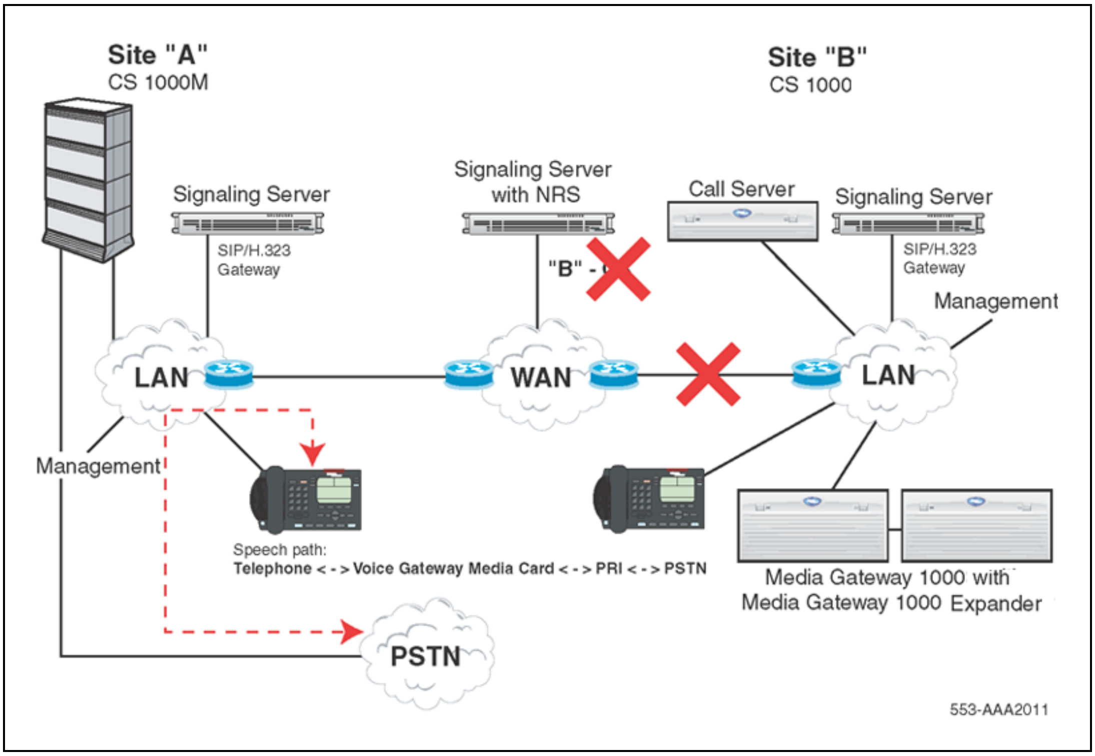
6. SIP/H.323 Gateway A replies to Call Server A with a message that all IP trunks are busy for the dialed DN. See Figure 47 "SIP/H.323 Gateway A replies to Call Server A" (page 109).

Figure 47
SIP/H.323 Gateway A replies to Call Server A



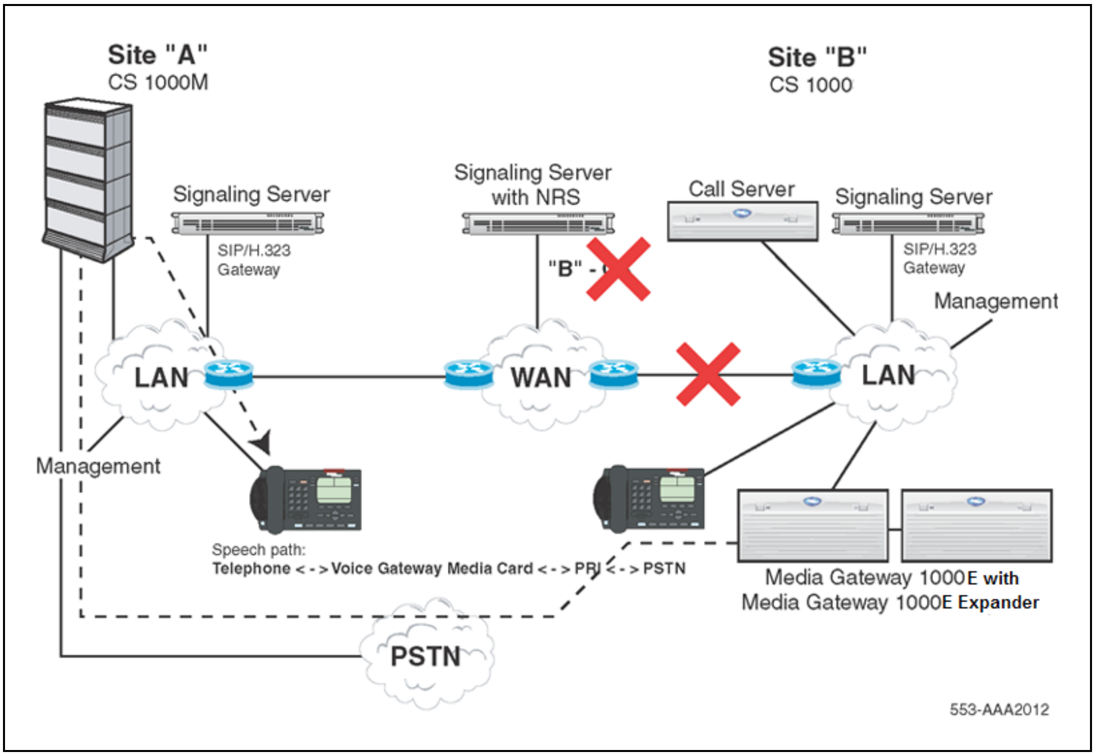
7. Call Server A chooses the next route in the Route List Data Block. The next route is a local PSTN trunk route. Call Server A allocates a Voice Gateway Media Card and PRI channel. Digit manipulation is applied to the route using the local PSTN. A successful call is made. See Figure 48 "Call Server A chooses the next route in the Route List Data Block" (page 110).

Figure 48
Call Server A chooses the next route in the Route List Data Block



8. The call is routed across PSTN and enables the users to talk to each other. The call is terminated over PSTN to Site B. See Figure 49 "Call is terminated over PSTN" (page 111).

Figure 49
Call is terminated over PSTN



Bandwidth Management

Contents

This section contains information on the following topics:

Introduction

CS 1000 supports Bandwidth Management on a network-wide basis so that voice quality can be managed between multiple Call Servers.

Bandwidth management allows for codec selection and bandwidth limitations to be placed on calls, depending on whether the calls are intrazone or interzone.

Adaptive Network Bandwidth Management is an enhancement of Bandwidth Management in which Quality of Service (QoS) metrics are used to automatically lower available bandwidth.

> **ATTENTION**
>
> **IMPORTANT!**
>
> Once all bandwidth is used, any additional calls are blocked or rerouted. Keep this in mind when designing and implementing Network Bandwidth Management.

Codec negotiation

Codec refers to the voice coding and compression algorithm used by DSPs. Each codec has different QoS and compression properties.

IP Peer Networking supports the per-call selection of codec standards, based on the type of call (interzone or intrazone). IP Peer Networking supports the following codecs (with supported payload sizes in parentheses, with the default value in bold):

- G.711 A/mu-law (10 ms, **20 ms**, and 30 ms)
- G.729 A (10 ms, **20 ms**, 30 ms, 40 ms, and 50 ms)
- G.729 AB (10 ms, **20 ms**, 30 ms, 40 ms, and 50 ms)
- G.723.1 (**30 ms**) (though it can limit the number of DSP channels available)
- T.38 for fax

  Note: The G.XXX series of codecs are standards defined by the International Telecommunications Union (ITU).

By default, the G.711 codec must be supported at both ends of a call. Codec configuration is performed for each node and is independent of the signaling gateway (SIP or H.323) that is used on the node.

If more than one codec is configured, then the minimum payload size among the configured codecs is used for the SIP Trunk Gateway codec negotiation.

> ***Note:*** Nortel recommends configuring the same payload size for all codecs in the same node. The payload size on the CS 1000 system must be set to 30 ms to work with the SRG.

SIP example

If a G.711 20 millisecond (ms) codec and G.729 30 ms codec are configured, then codec negotiation uses the minimum payload size of 20 ms. That is, the G.711 20 ms codec and the G.729 20 ms codec are used. Instead, Nortel recommends that both G.711 and G.729 codecs be configured as 20 ms.

Note: When a G.729 30 ms codec is configured, then the G.729 10 ms/20 ms/30 ms codecs are supported.

IP Peer Networking performs codec negotiation by providing a list of codecs that the devices can support. Use CS 1000 Element Manager to configure the list of codec capabilities. See Procedure 15 "Configuring codecs" (page 206) to configure codecs.

The codec preference sequence sent over SIP/H.323 depends on the bandwidth policy selected for the Virtual Trunk zone and the involved telephones. For "Best Quality", the list is sorted from best to worst voice quality. For "Best Bandwidth", the list is sorted from best to worst bandwidth usage.

The G.711 codec delivers "toll quality" audio at 64 kbit/s. This codec is optimal for speech quality, as it has the smallest delay and is resilient to channel errors. However, the G.711 codec uses the largest bandwidth.

The G.729A codec provides near toll quality voice at a low delay. The G.729A codec uses compression at 8 kbit/s. The G.729AB codec also uses compression at 8 kbit/s.

The G.723.1 codec provides the greatest compression.

Note 1: Payload default values need to be changed if the customer wants to communicate with a third-party gateway that does not support the above default payload sizes. Otherwise, IP Peer calls to or from the third-party gateway are not successful.

Note 2: If the payload sizes are set higher than the default values (for example, to support a third-party gateway), then the local IP calls are affected by higher latency. This is because the codec configuration applies to both IP Peer calls and local IP (IP Line) calls.

Note 3: If a CS1000E system is used, the same payload sizes for the same codec type should be configured on all IPMG cabinets in a system. Otherwise, TDM to TDM calls between IPMG cabinets are not successful.

G.711 A-law and mu-law interworking

In case the far end uses a different Pulse Code Modulation (PCM) encoding law for its G.711 codec, systems that are configured as G.711 A-law also include G.711 mu-law on their codec preferences list. Systems configured as G.711 mu-law include G.711 A-law as their last choice. Therefore, encoding law conversion is performed between systems with different laws.

Bandwidth management and codecs

Bandwidth management defines which codecs are used for intrazone call and interzone calls.

Bandwidth management enables administrators to define codec preferences for IP Phone-to-IP Phone calls controlled by the same CS 1000 system within the same zone. These calls are known as intrazone calls. This is different than the codec preferences for calls between an IP Phone on the CS 1000 system to a Virtual Trunk (potentially an IP Phone on another CS 1000 system) or calls to IP Phones is another zone. These calls are known as interzone calls.

For example, you may prefer high quality speech (G.711) over high bandwidth within one system, and lower quality speech (G.729 AB) over lower bandwidth to a Virtual Trunk. Such a mechanism can be useful when a system is on the same LAN as the IP Phones it controls, but the other systems are on a different LAN (connected through a WAN).

Virtual Trunks’ usage of bandwidth zones is different than IP Phone bandwidth usage. For Virtual Trunks, a zone number is configured in the Route Data Block (LD 16). The zone number determines codec selection for interzone and intrazone calls (that is, Best Bandwidth or Best Quality). See "Configuring IP Peer Networking" (page 171) .

Bandwidth usage for Virtual Trunks is accumulated in its zone to block calls that exceed the bandwidth availability in a specific zone. However, the amount of bandwidth that is required to complete a given call is not known until both call endpoints have negotiated which codec to use. The bandwidth used for calculating the usage of a Virtual Trunk call is determined by the preferred codec of the device that connects to the Virtual Trunk. If the device is an IP Phone, the bandwidth calculations use the preferred codec of the IP Phone, based on the codec policy defined for the zones involved (that is, Best Bandwidth or Best Quality). Likewise, the bandwidth calculations use the preferred codec of the Voice Gateway Media Card for connections between a circuit-switched device (for example, a PRI trunk) and a Virtual Trunk.

Codec selection

For every Virtual Trunk call, a codec must be selected before the media path can be opened. When a call is set up or modified (that is, media redirection), one of two processes occurs:

- The terminating node selects a common codec and sends the selected codec to the originating node.
- The codec selection occurs on both nodes.

Each node has two codec lists: its own list and the far end's list. In order to select the same codec on both nodes, it is essential to use the same codec selection algorithm on both nodes. Before the codec selection occurs, the following conditions are met:

- Each codec list contains more than one payload size for a given codec type (it depends on the codec configuration).
- Each codec list is sorted by order of preference (the first codec in the near end's list is the near end's most preferred codec, the first codec in the far end's list is the far end's preferred codec).

Codec selection algorithms

Once the codec lists meet the above conditions, one of the following codec selection algorithms selects the codec to be used:

- H.323 Master/Slave algorithm
- SIP Offer/Answer model
- "Best Bandwidth" codec selection algorithm

Note: If a SIP trunk call is between a CS 1000 system and other third-party gateway/SIP clients (for example, MCS 5100), then the codec selection does not guarantee that the same codec is selected for a call from endpoint A to endpoint B and for a call from endpoint B to endpoint A. This different codec selection makes it difficult for bandwidth management. However, calls between two CS 1000 systems have the same codec selection decision regardless of who originated the call.

H.323 Master/Slave algorithm

In the case of a Virtual Trunk call between Nortel and third-party equipment, the H.323 Master/Slave algorithm is used.

The codec selection algorithm proposed by the H.323 standard involves a Master/Slave negotiation. This is initiated each time two nodes exchange their capabilities (TCS message). The Master/Slave information decides that one node is Master and the other node is Slave. The outcome of the Master/Slave negotiation is not known in advance; it is a random result. One node could be Master then Slave (or vice versa) during the same call.

Algorithm details The H.323 Master/Slave algorithm operates in the following manner:

- The Master node uses its own codec list as the preferred one and finds a common codec in the far end's list. In other words, the Master gets the first codec in its list (for example, C1), checks in the far end's list if it is a common codec; if it is, C1 is the selected codec. Otherwise, it gets the second codec in its list and verifies it against the far end, and so on.

- The Slave node uses the far end's list as the preferred one and finds in its own list the common codec.

Issues caused by the H.323 Master/Slave algorithm The issues caused by the Master/Slave algorithm are due to the random nature of the Master/Slave information. In other words, one cannot predetermine the codec that is used during a Virtual Trunk call.

The following are the issues associated with the H.323 Master/Slave algorithm:

- After an on-hold and off-hold scenario (which triggers Master/Slave negotiation), the codec used for the restored call might be different than the one used before on-hold, because the Master/Slave information could have been changed.
- When using "Fast Start" codec selection, a call from Telephone 1 (node1) to Telephone 2 (node2) can use a different codec than a call from Telephone 2 (node2) to Telephone 1 (node1), because the terminating end is always Master.
- For tandem calls, the Master/Slave information is not relevant. The Master/Slave information is designed for use between two nodes only, not between three or more nodes. It makes the codec selection for tandem calls more complex and inefficient.

To solve the issues, another codec selection algorithm, not based on the unpredictable Master/Slave information, is needed. Since any change to the Master/Slave algorithm implies a change to the H.323 standard, the new codec algorithm is used for Virtual Trunk calls between Nortel equipment.

SIP Offer/Answer model

The SIP codec negotiation is based on the Offer/Answer model with Session Description Protocol (SDP).

The following three cases of codec negotiation are supported:

- The calling user agent sends an SDP offer with its codec list in the INVITE message with a *sendrecv* attribute. In this case, the called user agent selects one codec and sends the selected codec in an SDP answer. The SDP answer is included in the 200 OK message (which is the response to the INVITE) with the *sendrecv* attribute.

  This is the preferred method of operation.
- The calling user agent sends an SDP offer with its codec list in the INVITE message with a *sendrecv* attribute. The called user agent returns more than one codec in the SDP answer. In the case that many codecs are included in the response, the calling user agent picks the first compatible codec from the called user agent's list, and sends a new SDP offer with a single codec to lock it in.

- If the SDP of the calling user agent is not present in the INVITE message, then the called user agent sends its codec list in an SDP offer in the 200 OK message, with the *sendrecv* attribute. The calling user agent selects one codec and sends the selected codec in an SDP answer inside the *ACK* message, with *sendrecv* attribute.

For more information on this algorithm, refer to RFC 3264 – An Offer/Answer Model with the Session Description Protocol (SDP).

Best Bandwidth codec selection algorithm

The "Best Bandwidth" codec selection algorithm solves the issues caused by the H.323 Master/Slave algorithm. The "Best Bandwidth" algorithm selects one common codec based on two codec lists. Every time the selection is done with the same two lists, the selected codec is the same.

The "Best Bandwidth" codec decision is based on the codec type only, it does not take into account the fact that some codecs, while generally using less bandwidth, can consume more bandwidth than others at certain payload sizes.

"Best Bandwidth" is also applicable to SIP.

Algorithm details The selected codec is the type considered as the best bandwidth codec type. To know whether one codec type has better bandwidth than another, see the rule as summarized in Table 8 "Best Bandwidth algorithm — codec type" (page 119).

Table 8
Best Bandwidth algorithm codec type

	G.711 A law	**G.711 mu-law**	**G.729 A**	**G. 729 AB**	**G. 723.1**
G.711 A-law	G.711 A-law	G.711 mu-law	G.729 A	G. 729 AB	G. 723.1
G.711 mu-law	G.711 mu-law	G.711 mu-law	G.729 A	G. 729 AB	G. 723.1
G.729 A	G.729 A	G.729 A	G.729 A	G. 729 AB	G.729 A
G. 729 AB	G. 729 AB	G. 729 AB	G. 729 AB	G. 729 AB	G. 729 AB
G. 723.1	G. 723.1	G. 723.1	G.729 A	G. 729 AB	G. 723.1

Interoperability between CS1000 and SRG

The SRG is designed to interoperate with this feature in a manner similar to MG 1000B, but with a limitation with respect to codec selection policy.

Calls between branch IP Phones and the branch PSTN or between branch IP Phones and branch analog phones are based on the interzone policy rather than the intrazone policy defined in the CS 1000 main office. The zone table is updated based on the intrazone policy.

The net result of this limitation is that calls between branch IP Phone users and the branch PSTN, or between branch IP Phones and branch analog telephones, always use a Best Bandwidth codec. However, the calls are accounted for as Best Quality. This can impact the perception of call quality in this scenario, but it does not result in early call blocking. There are no impacts to codec selection or bandwidth usage tracking for calls that require WAN bandwidth.

Configuring Bandwidth Management

The following sections describe how to configure Bandwidth Management in a CS 1000 network. Nortel recommends that you read the Bandwidth Management section in *Converging the Data Network with VoIP* (NN43001-260) before using the following configuration information.

Zones

Bandwidth Management Zones are configured for each endpoint on a Call Server. The Network Bandwidth Zone number determines if a call is an intrazone call or an interzone call. Once that is determined, the proper codec and bandwidth limit is applied to the call.

All of the endpoints on one Call Server are configured with Zone number to identify all of the endpoints as being in a unique geographic location in the network. In addition, Virtual Trunks are configured with a Zone number that is different from the endpoint Zone numbers in the Call Server.

Codec selection occurs as described in "Codec selection" (page 116).

Configuration rules

There are three configuration rules for Bandwidth Management, as follows:

1. Each Call Server in the network must be configured with a unique VPNI, with the only exception noted in point 2, next.
2. Branch office (MG 1000B and SRG) Call Servers must be configured with the same VPNI as that of the main office Call Server with which they register.
3. Virtual Trunks must be configured with a different Zone number than the endpoints.

Network Planning

Before configuring Bandwidth Management in a CS1000 network, follow these steps:

Step	Action
1	Choose unique VPNIs for all Call Servers in the network.

2 Choose unique Bandwidth Zone numbers for all Call Servers in the network. These are used when configuring the endpoints (telephones and gateways) on the Call Server.

3 Choose unique Bandwidth Zone numbers for the Virtual Trunks in the network.

4 Choose the codecs that will be enabled on each Call Server.

5 Identify what the interzone codec strategy will be Best Bandwidth (BB) or Best Quality (BQ) for each zone in the network.

6 Identify what the intrazone codec strategy will be Best Bandwidth (BB) or Best Quality (BQ) for each zone in the network.

7 Calculate the bandwidth available for intrazone calls for each zone in the network.

8 Calculate the bandwidth available for interzone calls for each zone in the network.

9 Calculate the bandwidth available for intrazone calls

—End—

Enabling codecs

In Element Manager, select the codecs that will be enabled for calls on the Call Server, and define the associated parameters, such as payload size. Use Procedure 15 "Configuring codecs" (page 206) to view available codecs, and configure existing or new codecs.

Configuring Bandwidth Management parameters

The steps to configure Bandwidth Management on the Call Server are as follows:

Step Action

1 Define a VPNI number in LD 15.

2 Configure the Bandwidth Management parameters for each zone on the Call Server using either Element Manager (see "Configuration using CS 1000 Element Manager" (page 122)) or LD 117 (see "Configuration using LD 117" (page 123)):

- Call Server zones that will be used for endpoints (telephones and gateways) with the following properties:
  - — Intrazone Preferred Strategy = Best Quality (BQ)

- Intrazone Bandwidth = default (1000000)
- Interzone Preferred Strategy = Best Bandwidth (BB)
- Interzone Bandwidth = maximum bandwidth usage allowed between peer Call Servers

- Call Server zones that will be used for Virtual Trunks with the following properties:
  - Intrazone Preferred Strategy = Best Quality (BQ)
  - Intrazone Bandwidth = default (1000000)
  - Interzone Preferred Strategy = Best Bandwidth (BB)
  - Interzone Bandwidth = default (1000000)

3 Configure the IP Phone, DSP and Virtual Trunk data with the corresponding zone numbers.

For example, for an IP Phone 2004 telephone in zone 8:

```
LD 11
REQ NEW
TYPE 2004p2
...
ZONE 8
...
```

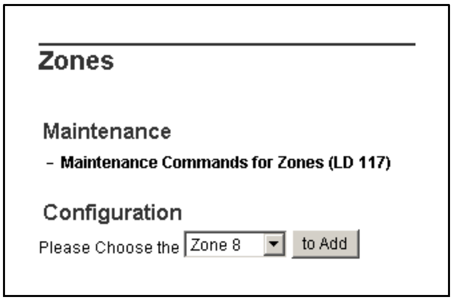
—End—

Configuration using CS 1000 Element Manager

Zones are configured from the Zones web page, shown in Figure 50 "Zones web page" (page 122).

Use Procedure 8 "Configuring zones" (page 187) to configure a Network Bandwidth Management zone, using the values given on "Configuring Bandwidth Management parameters" (page 121).

Figure 50
Zones web page



Zones

Maintenance

- Maintenance Commands for Zones (LD 117)

Configuration

Please Choose the Zone 8 to Add

Configuration using LD 117

A new Bandwidth Management zone is configured in LD 117 using the NEW ZONE command. An existing zone can be modified using the CHG ZONE command.

LD 117 Configure a new or existing Bandwidth Management zone.

Command	Description
NEW | CHG ZONE <zoneNumber> [<intraZoneBandwidth> <intraZoneStrategy> <interZoneBandwidth> <interZoneStrategy> <zoneIntent> <zoneResourceType>]	
	Configure a new zone (NEW) or change (CHG) an existing zone, where: • zoneNumber = 0-255 • intraZoneBandwidth = Available intrazone bandwidth (Kbit/s); Nortel recommends this value be set to the maximum value. • intraZoneStrategy = BB (Best Bandwidth) or BQ (Best Quality); Nortel recommends this value be set to BQ. • interZoneBandwidth = — For Call Server zone = Maximum bandwidth usage (in Kbit/s) allowed between peer Call Servers — For Virtual Trunk zones = 1000000 (Kbit/s) • interZoneStrategy = BB (Best Bandwidth) or BQ (Best Quality); Nortel recommends this value be set to BB to conserve interzone bandwidth. • zoneIntent = type of zone, where: — MO = Main office (Call Server) zone — BMG = Branch Media Gateway (for branch office zones) — VTRK = Virtual Trunk zone • zoneResourceType = resource intrazone preferred strategy, where: — shared = shared DSP channels (default) — private = private DSP channels ***Note:*** In CS 1000 Release 4.5, the zones that were described with BMG designator stay with BMG one, all the other zones are provided with MO designator. It is possible to update ZoneIntent using CHG ZONE command.

Maintenance commands

Maintenance commands can be run from Element Manager or LD 117.

Element Manager (EM) for CS 1000 Release 5.0 is offered in two versions: a Linux-based EM and a VxWorks-based EM. The Linux-based EM is hosted in a stand-alone mode on a dedicated server running the Linux™ real-time operating system. The VxWorks-based EM is hosted co-resident with Signaling Server applications on a server running the VxWorks™ real-time operating system.

Maintenance using Element Manager

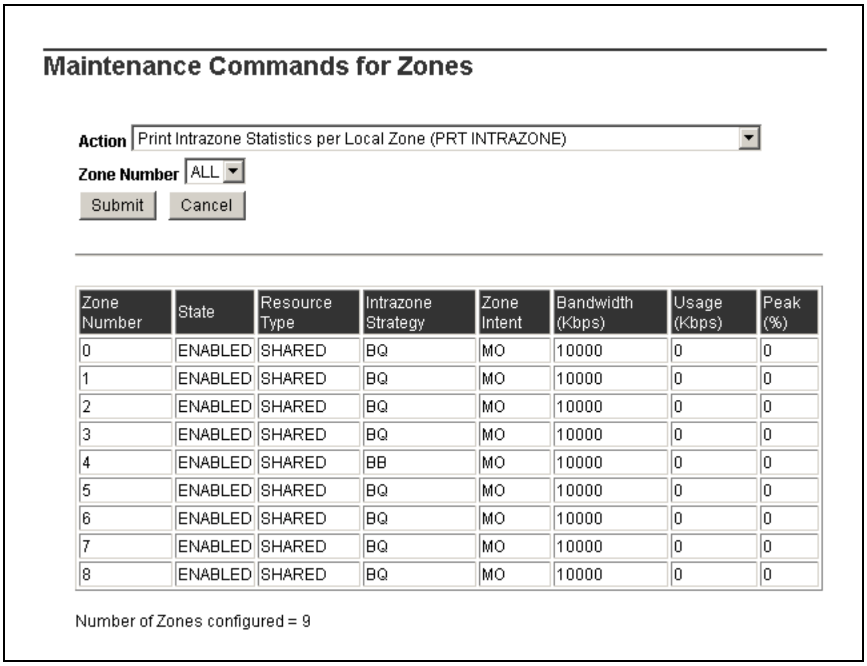
The PRT INTRAZONE and PRT INTERZONE commands are available in Element Manager from the Zones web page, shown in Figure 50 "Zones web page" (page 122). To access these commands, follow the steps in Procedure 1 "Printing intrazone and interzone statistics for a zone" (page 124).

Procedure 1

Printing intrazone and interzone statistics for a zone

Step	Action
1	In the EM Navigator select **System > IP Network > Zones**. The **Zones** web page opens, as shown in Figure 50 "Zones web page" (page 122).
2	Click **Maintenance Commands for Zones (LD 117).** The **Maintenance Commands for Zones** web page opens, as shown in Figure 51 "Maintenance Commands for Zones web page" (page 125). This page lists all the configured zones.

Figure 51
Maintenance Commands for Zones web page



Maintenance Commands for Zones

Action: Print Intrazone Statistics per Local Zone (PRT INTRAZONE)

Zone Number: ALL

Zone Number	State	Resource Type	Intrazone Strategy	Zone Intent	Bandwidth (Kbps)	Usage (Kbps)	Peak (%)
0	ENABLED	SHARED	BQ	MO	10000	0	0
1	ENABLED	SHARED	BQ	MO	10000	0	0
2	ENABLED	SHARED	BQ	MO	10000	0	0
3	ENABLED	SHARED	BQ	MO	10000	0	0
4	ENABLED	SHARED	BB	MO	10000	0	0
5	ENABLED	SHARED	BQ	MO	10000	0	0
6	ENABLED	SHARED	BQ	MO	10000	0	0
7	ENABLED	SHARED	BQ	MO	10000	0	0
8	ENABLED	SHARED	BQ	MO	10000	0	0

Number of Zones configured = 9

3 Do one of the following:

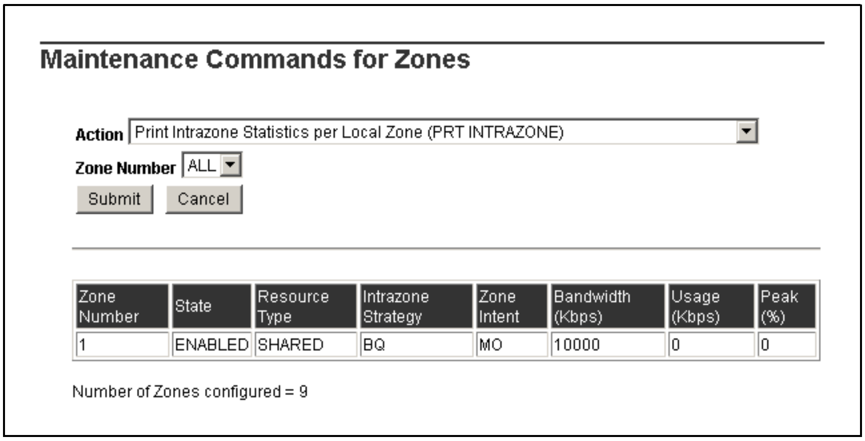
- To display interzone statistics:
  1. Select **Print Interzone Statistics (PRT INTERZONE)** from the **Action** drop-down list.
  2. Select a zone from the **Zone Number** drop-down list, by doing one of the following:
     - Select **ALL** to print statistics for all zones.
     - Select a specific zone number to display statistics for a specific zone.
- To display intrazone statistics:
  1. Select **Print Intrazone Statistics per Local Zone (PRT INTRAZONE)** from the **Action** drop-down list.
  2. Select a zone from the Near End Zone Number drop-down list, by doing of the following:
     - Select **ALL** to print statistics for all zones.
     - Select a specific zone number to display statistics for a specific zone.

4 Click **Submit**.

The **Maintenance Commands for Zones** web page reopens, displaying the statistics for the specified zone or zones. A blank field indicates that the statistic is either not available or not applicable to that zone.

Figure 52 "Element Manager — intrazone statistics" (page 126) shows an example of intrazone statistics for a sample Zone 1. Figure 53 "Element Manager — interzone statistics" (page 126) shows an example of interzone statistics for the same Zone 1.

Figure 52
Element Manager intrazone statistics



Maintenance Commands for Zones

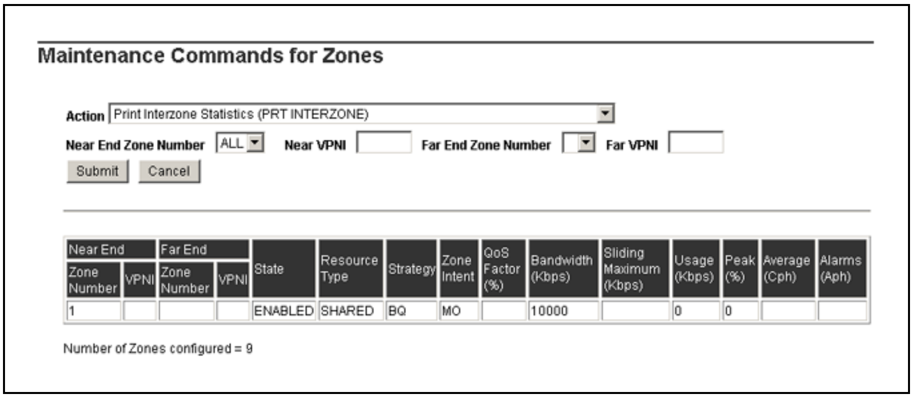
Action: Print Intrazone Statistics per Local Zone (PRT INTRAZONE)

Zone Number: ALL

Zone Number	State	Resource Type	Intrazone Strategy	Zone Intent	Bandwidth (Kbps)	Usage (Kbps)	Peak (%)
1	ENABLED	SHARED	BQ	MO	10000	0	0

Number of Zones configured = 9

Figure 53
Element Manager interzone statistics



Maintenance Commands for Zones

Action: Print Interzone Statistics (PRT INTERZONE)

Near End Zone Number: ALL Near VPNI: Far End Zone Number: Far VPNI:

Near End		Far End		State	Resource Type	Strategy	Zone Intent	QoS Factor (%)	Bandwidth (Kbps)	Sliding Maximum (Kbps)	Usage (Kbps)	Peak (%)	Average (Cph)	Alarms (Aph)
Zone Number	VPNI	Zone Number	VPNI											
1				ENABLED	SHARED	BQ	MO		10000		0	0		

Number of Zones configured = 9

—End—

Maintenance using LD 117

Use the PRT INTRAZONE or PRT INTERZONE commands in LD 117 to view the intrazone or interzone statistics for specified zones.

Use the PRT ZQOS / PRT AQOS commands in LD 117 to display QoS records for zones.

> ***Note:*** Do not use the PRT ZONE command — it has been replaced by the PRT INTRAZONE and PRT INTERZONE commands.

LD 117 Print zone statistics.

Command	Description
PRT INTRAZONE {<zone>}	Print intrazone statistics for the identified zones, where: • zone = ALL or 0-255 The output of this command displays the following information: • Zone • Type = PRIVATE/SHARED • Strategy = BB/BQ • ZoneIntent = MO/BMG/VTRK • Bandwidth = number of Kbps • Usage = number of Kbps • Peak = %
PRT INTERZONE {<nearZone>} [{<nearVPNI>} {<farZone>} {<farVPNI>}]	Print interzone statistics for the specific VPNI zone; where: • nearZone = ALL or 0-255 The output of this command displays the following information: • Zone number = 0-255 • Zone VPNI = 1-16283 • Type= PRIVATE/SHARED • Strategy = BB/BQ • ZoneIntent = MO/VTRK
PRT AQOS <attribute> <zone> PRT AQOS <attribute> ALL	

Command	Description
PRT ZQOS <zone> <attribute> PRT ZQOS <zone> ALL	Prints QoS records for specified attribute and zone (or for all zones with ALL). Where <attribute> is the attribute [1-32] as defined in the Traffic Report 16 (see *Traffic Measurement: Formats and Output (NN43001-750)*), for example "Interzone warning jitter count.
	Prints QoS records for specified attribute and zone (or for all attributes with ALL). Where <attribute > is the attribute as defined in the Traffic Report 16 (see *Traffic Measurement: Formats and Output (NN43001-750)*), for example "Interzone warning jitter count".

Adaptive Network Bandwidth Management

Description

The Adaptive Network Bandwidth Management feature enhances the performance of Voice over Internet Protocol (VoIP) networks based on real-time interaction. It provides the means to automatically adjust bandwidth limits and take corrective action in response to Quality of Service (QoS) feedback. This dynamic bandwidth adjustment maintains a high level of voice quality during network degradation.

The Adaptive Network Bandwidth Management feature dynamically adapts to QoS in the network and reduces the bandwidth available for interzone calls if QoS degrades. Typically, each Call Server in the network has a zone assigned to it. The Call Server keeps track of the bandwidth being used between its own zone and zones belonging to other Call Servers. If the QoS degrades between the Call Server zone and a zone belonging to another Call Server, the available bandwidth is reduced automatically between those two zones. When the QoS between the two zones improves, then the bandwidth limit is allowed to return to normal.

When an IP Phone encounters degradation of the network, it informs the Call Server through various QoS alarms. These QoS alarms (packet loss, jitter, delay, and, for phase 2 IP Phones, R value) get reported to the Call Server. Depending upon the rate of the incoming alarms and the value of the alarms, the Call Server reduces the available bandwidth available to make new calls. The Call Server will lower/limit the number of new calls allowed, based on the available bandwidth. This prevents excessive calls being placed on a network with limited bandwidth (resulting in poor voice quality). Once the adjusted (lowered) bandwidth reaches its full capacity, new calls are either routed to an alternate route (if available) using Network Alternate Routing Service (NARS) or the Alternative Routing for NBWM feature (see *Branch Office Installation and Commissioning* (NN43001-314), or new calls are blocked. The Call Server continues to monitor the network throughout the network degradation period. When the degradation is removed or the performance of the network improves, the allowable bandwidth returns to provisioned levels and the Call Server gradually starts allowing new calls.

Essentially, Adaptive Network Bandwidth Management provides a fallback to PSTN on QoS degradation for new calls. As a result, bandwidth is managed and quality measured between all the zones across the entire network, and when necessary corrective action is taken. Due to the real-time interaction with the network, less maintenance is required for the network since the system reacts automatically to network conditions.

With Adaptive Network Bandwidth Management, it is not necessary to provision bandwidth parameters between every zone in the network. Rather, the Call Server automatically learns of new zones in the network and applies Adaptive Network Bandwidth Management to these new zones as required. Therefore, as new Call Servers are added to the network, it is not necessary to re-provision all the other Call Servers on the network to take into account this new Call Server. Conversely, when Call Servers are removed from the network, the remaining Call Servers age out the old Call Server information and therefore, provide only up to date bandwidth information.

This feature operates between all IP Peer CS 1000 systems, including the Media Gateway 1000B and Survivable Remote Gateway 50.

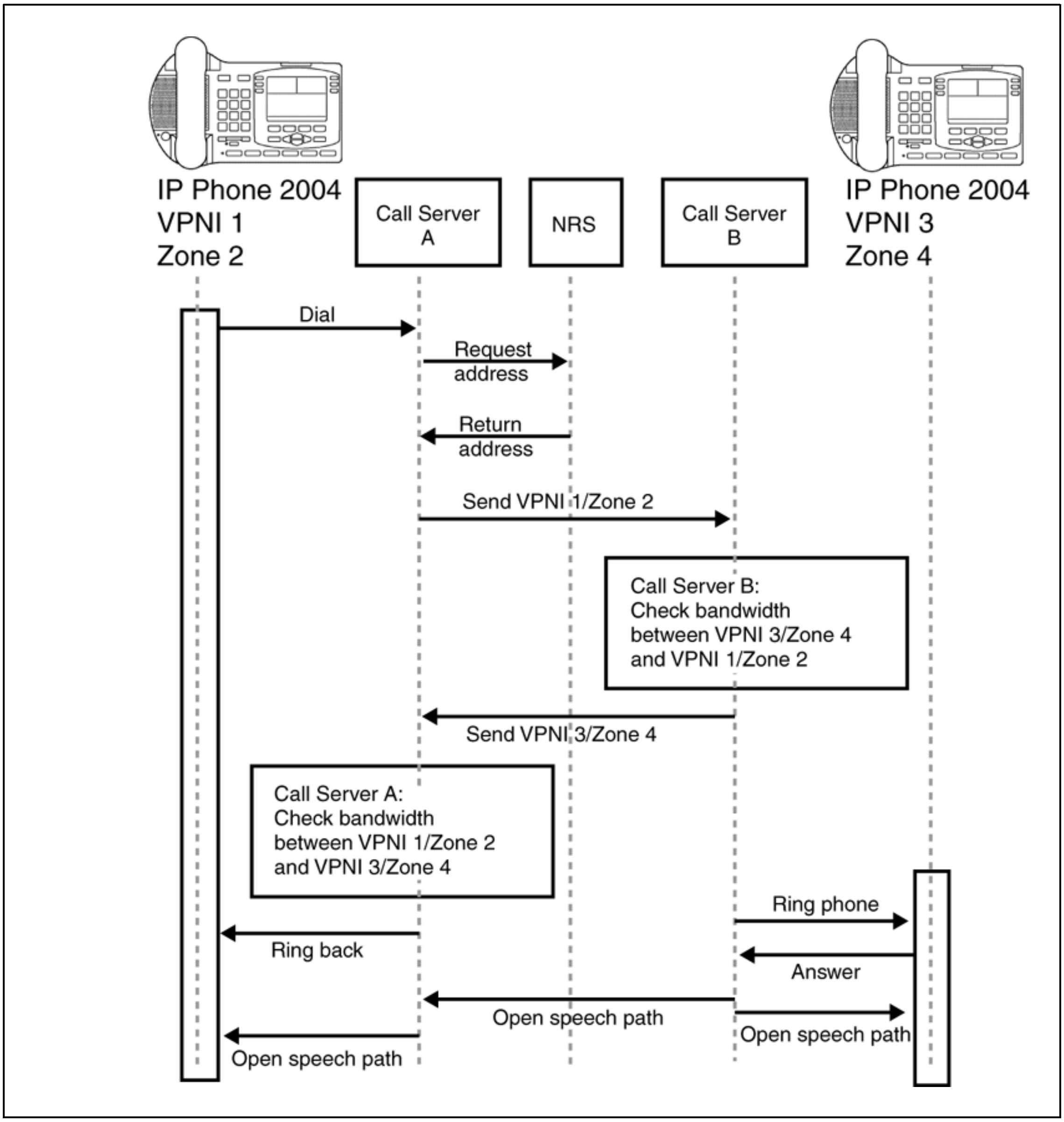
Call scenario

A call is requested from a telephone in VPNI 1/Zone 2 on Call Server A to a telephone in VPNI 3/Zone 3 on Call Server B. Both zones have Adaptive Network Bandwidth Management enabled.

1. Call Server A contacts the Network Redirect Server to obtain the address of Call Server B.
2. Call Server A sends a call setup message to Call Server B, identifying the calling telephone’s VPNI and zone.
3. Call Server B determines if there is sufficient bandwidth for the call, and sends back the VPNI and zone of the called telephone.
4. Call Server A checks its bandwidth table to determine if there is sufficient bandwidth available for the call from Call Server A to Call Server B.
5. If Call Server A determines there is enough bandwidth available, the call is established; otherwise, alternate treatment is provided in the form of blocking or rerouting the call.

Both Call Server A and Call Server B must consult their own bandwidth tables to determine if there is enough bandwidth for the call to proceed. Figure 54 "Call Progress with Adaptive Network Bandwidth Management" (page 130) shows this scenario.

Figure 54
Call Progress with Adaptive Network Bandwidth Management



Zone bandwidth management and Adaptive Network Bandwidth Management

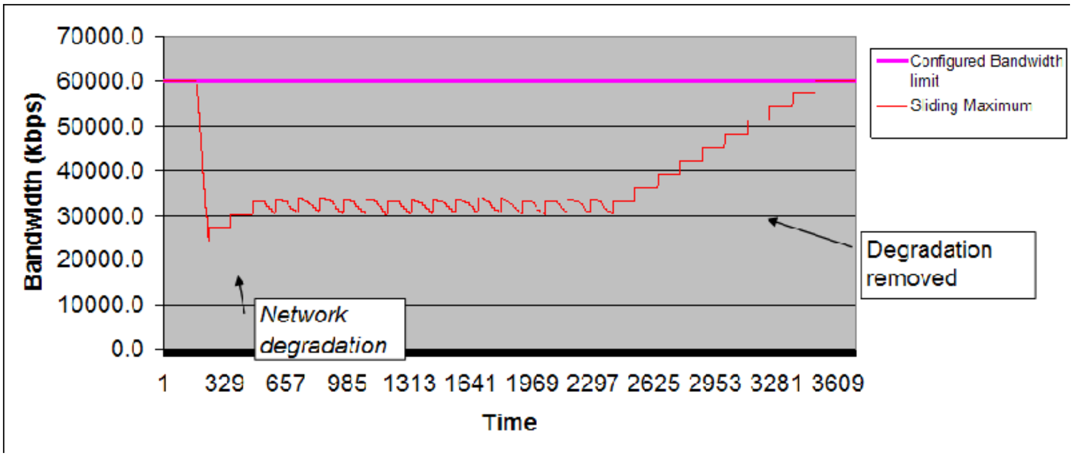
Using Element Manager or the Command Line Interface (CLI), previously configured zones (except Zone 0) can have the Adaptive Network Bandwidth Management feature turned on or off. Once turned on, alarm threshold levels and the QoS coefficients can be adjusted from the default values. Adaptive Network Bandwidth Management cannot be enabled for Zone 0.

When Adaptive Network Bandwidth Management is enabled for a particular zone on the Call Server, the zone appears in the zone table. The zone table can be displayed using Element Manager or LD 117. When a call is made from the configured zone to another zone, the bandwidth used appears in the zone table.

When a call is made from a zone with Adaptive Network Bandwidth Management enabled, to a third party gateway, which has no zone, then the zone of the Virtual Trunk (VTRK) is used and appears in the zone table.

Figure 55 "Adaptive Network Bandwidth Management graph" (page 131) shows an example of the bandwidth changes.

Figure 55
Adaptive Network Bandwidth Management graph



When a Call Server receives a QoS alarm, the two zones that originated the alarm are determined. Using this information, the Call Server reduces the bandwidth limit between the two zones. This zone-to-zone bandwidth limit (in effect at any particular time) is known as the Sliding Maximum Bandwidth Limit and is a percentage of the Configured Interzone bandwidth limit. The Sliding Maximum Bandwidth Limit value is displayed using the command `prt interzone` command. The `QoS Factor %` value, also displayed by this command, is a ratio of the Sliding Maximum Bandwidth

Limit and the configured allowable bandwidth expressed as a percentage. The Call Server checks the Network Bandwidth zone management tables for the originating and terminating zones of the new call to determine the available bandwidth for the call.

For more information about alarms, refer to *Software Input/Output: System Messages* (NN43001-712).

When feedback indicates a significant QoS change in a zone, the Call Server reduces the available bandwidth (Sliding Maximum Bandwidth Limit) in the zone until the QoS reaches a satisfactory level. Once satisfactory QoS is reached, the bandwidth is slowly raised until either the full bandwidth is available or until QoS degrades again. Bandwidth changes can be configured to be gradual (to reduce rapid swings and variations) or rapid.

Multiple Appearance Directory Numbers (MADN) can exist on different zones. Calls to an MADN are handled the same as other IP Phone calls, and are subject to the same bandwidth limitations.

New SNMP alarms are provided to monitor the system. When the bandwidth limit between zones is reduced below configured levels, an alarm is raised. A Warning alarm and an Unacceptable alarm, each corresponding to a drop below a configured threshold, are used. When the bandwidth returns to normal, the alarm is cleared. If the bandwidth limit reaches zero, an additional Unacceptable alarm is raised. These alarms allow the system administrator to monitor the system and take corrective action when required.

Adaptive Network Bandwidth Management configuration parameters

Packet Loss (pl), Jitter (j) and Delay (d) measurements, along with the R factor (r) in IP Phone 200x Phase II telephones, are used to calculate the QoS level for the zones. The coefficients for these QoS measurements — packet loss (Cpl), jitter (Cj), delay (Cd), and the R factor (Cr) — can be configured and are used to calculate the rate of bandwidth change. Increasing them from their default values causes the Sliding Maximum to decrease faster in response to the specific QoS alarm.

The QoS Coefficient (CQoS) can be varied from its default value. Increasing this value causes the Sliding Maximum to change more rapidly in response to QoS alarms. However, making this value too large will result in loss of overall bandwidth, as shown in Figure 56 "Effect of the default CQos Coefficient" (page 133) below and Figure 57 "Effect of a higher CQoS Coefficient" (page 133).

Figure 56
Effect of the default CQos Coefficient

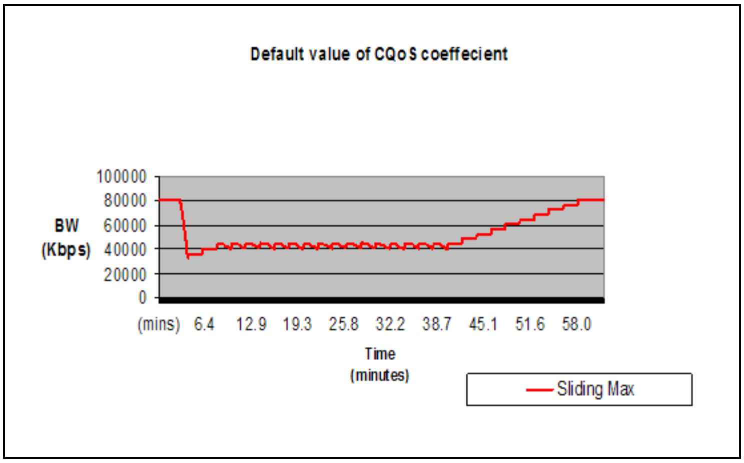
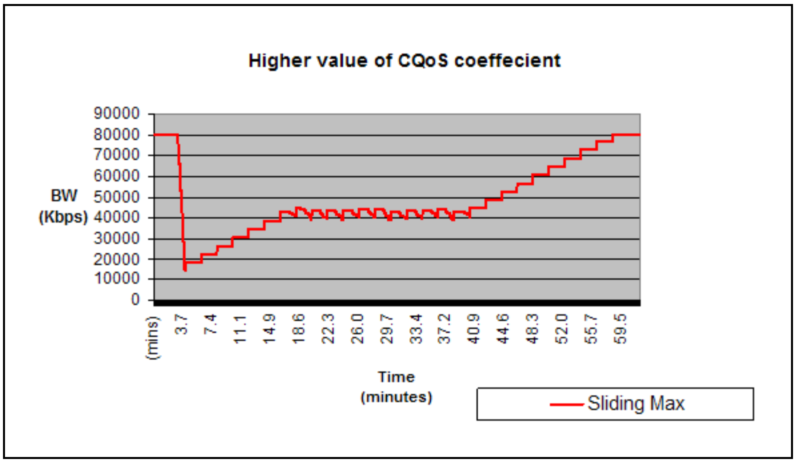


Figure 57
Effect of a higher CQoS Coefficient



Other configurable coefficients used in the calculation are the QoS Coefficient (CQoS), QoS Response Time Increase (ZQRT), and QoS Response Time Interval (ZQRTI). CQoS, Cr, Cd, Cpl, and Cj control the rate of bandwidth decrease, while ZQRT and ZQRTI control the rate of bandwidth increase.

The Call Admission Control (CAC) Validity Time Interval (CACVT) is used to control the length of time that records from a Call Server are saved in the Bandwidth Management table. If no calls occur between two Call Servers within the configured time, the Call Server is removed from the table. For example, if Call Server A has Call Server B in the table, and no call is placed between A and B for the CACVT time, then Call Server A removes all Call Server B records in the table.

Limitations

Virtual Office IP Phones are not subject to bandwidth limitations. They may not have the correct zone information configured. They can also be controlled by a Call Server that is not responsible for the particular zone. Thus, bandwidth management is not possible for these phones.

Feature packaging

The Adaptive Network Bandwidth Management feature requires the following packages:

- QoS Enhanced Reporting (PVQM) package 401

  Note: Package 401, QoS Enhanced Reporting (PVQM), is required if the R value from the Phase II IP Phones is to be reported and used in the calculations.

- Call Admission Control (CAC) package 407

Configuration rules

The configuration rules for Adaptive Network Bandwidth Management are as follows:

- Each main office Call Server in a network must have a unique VPNI (which cannot be zero).
- All branch offices (MG1000B or SRG) associated with a particular main office must have the same VPNI as the main office Call Server.
- All IP Phones (other than specific IP SoftPhone 2050s) and DSP endpoints on a Call Server must be configured for the same zone.
- IP SoftPhone 2050s being used remotely must be configured for Zone 0.
- Branch office systems (MG 1000B or SRG) should tandem all calls through the main office Call Server to allow bandwidth monitoring and control. In this case, the media path is direct between the branch office and any point in the network.
- Trunk Route Optimization (TRO) must be disabled between the main office Call Server and the MG 1000B Core or SRG. In this case, the media path is direct between the branch office and any point in the network.
- Adaptive Network Bandwidth Management parameters are configured on the main office only and must not be configured at the branch offices.

Configuring Adaptive Network Bandwidth Management

The following is a summary of the tasks necessary to configure Adaptive Network Bandwidth Management in the network.

1. Enable the Call Admission Control (CAC) package.

2. Configure CAC in Element Manager or LD 117:
    a. Configure the VPNI on the main office and branch offices.
    b. Configure both the main office and branch office zones at the main office.
    c. Configure the branch office zone on the MG 1000B Core or SRG.
    d. Configure the interzone and intrazone bandwidth limits at the main office and MG 1000B Core or SRG.
    e. Enable Adaptive Network Bandwidth Management for the zones on the main office Call Server.
    f. If required, alter the Adaptive Network Bandwidth Management parameters in keeping with the information in "Advanced Configuration Notes" (page 135) below.
3. Tandem the outbound branch office calls by configuring the NRS.
4. Tandem the inbound branch office calls by creating a dialing plan which routes all calls destined for the branch office through the main office.

Advanced Configuration Notes

1. The default values for Cpl, Cj, Cd, Cr and CQos can be increased to increase the response time for Sliding Maximum changes. However, increasing them can cause the Sliding Maximum to temporarily decrease to a lower value then necessary, resulting in the needless blocking of interzone calls.
2. Increasing the value of ZQRT will increase the speed at which the Sliding Maximum increases. The same effect can be achieved by decreasing ZQRTI. However, changing these values can cause the Sliding maximum to oscillate until the network degradation is removed.
3. It may be necessary to change the notification level (ZQNL) of the Call Server so it can react to the QoS alarms. Use LD 117 to change this level. Refer to *Converging the Data Network with VoIP (NN43001-260)* for information on notification levels for alarms.

Configuration using Element Manager

Element Manager can be used to enable and configure the feature.

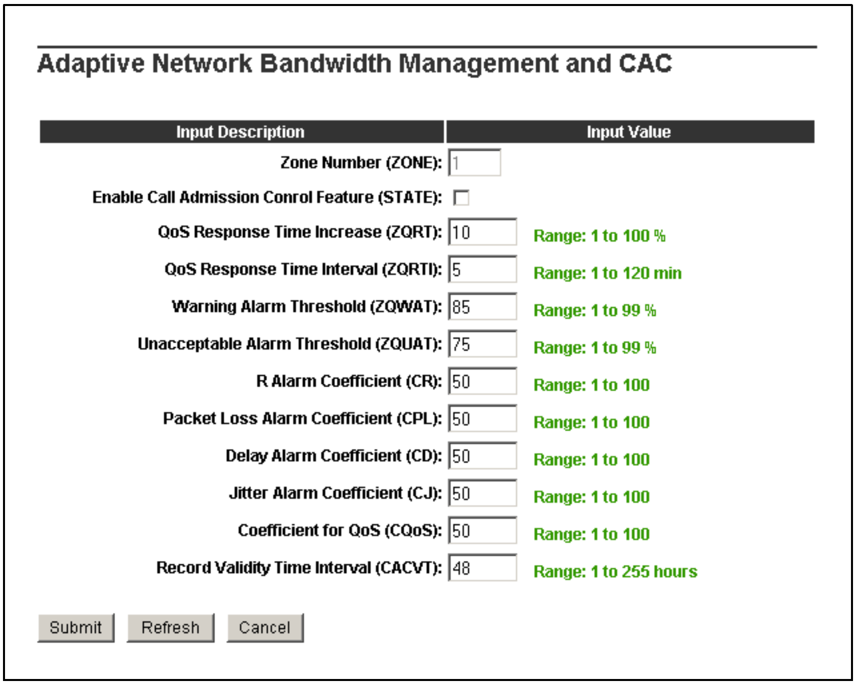
The zone must exist before it can be configured for Adaptive Network Bandwidth Management. Use Procedure 8 "Configuring zones" (page 187) to create and configure the basic properties of the zone.

To configure the Adaptive Network Bandwidth Management feature, select a zone on the Zones web page (see Figure 50 "Zones web page" (page 122)) and click **Adaptive Network Bandwidth Management and CAC**.

The **Adaptive Network Bandwidth Management and CAC** web page opens, as shown in Figure 58 "Adaptive Network Bandwidth Management and CAC web page" (page 136).

> ***Note:*** Do not configure Adaptive Network Bandwidth Management for Zone 0 or Virtual Trunk zones.

Figure 58
Adaptive Network Bandwidth Management and CAC web page



Adaptive Network Bandwidth Management and CAC

Input Description	Input Value	
Zone Number (ZONE):	1	
Enable Call Admission Conrol Feature (STATE):	☐	
QoS Response Time Increase (ZQRT):	10	Range: 1 to 100 %
QoS Response Time Interval (ZQRTI):	5	Range: 1 to 120 min
Warning Alarm Threshold (ZQWAT):	85	Range: 1 to 99 %
Unacceptable Alarm Threshold (ZQUAT):	75	Range: 1 to 99 %
R Alarm Coefficient (CR):	50	Range: 1 to 100
Packet Loss Alarm Coefficient (CPL):	50	Range: 1 to 100
Delay Alarm Coefficient (CD):	50	Range: 1 to 100
Jitter Alarm Coefficient (CJ):	50	Range: 1 to 100
Coefficient for QoS (CQoS):	50	Range: 1 to 100
Record Validity Time Interval (CACVT):	48	Range: 1 to 255 hours

If the Adaptive Network Bandwidth Management feature is enabled using the **Enable Call Admission Control feature (ZCAC)** check box, then the other parameters can be adjusted as required.

Table 9 "Adaptive Network Bandwidth Management and CAC fields" (page 136) shows the fields in the **Adaptive Network Bandwidth Management and CAC** web page, the field definitions, and their LD 117 command equivalent.

Table 9
Adaptive Network Bandwidth Management and CAC fields

Field Title	Field Definition	LD 117 equivalents
Enable Call Admission Control Feature (CAC)	Control the CAC feature for the zone • Enable (check box selected)	ENL ZCAC DIS ZCAC

Field Title	Field Definition	LD 117 equivalents
	• disable (clear the check box)	
QoS Response Time Increase (ZQRT)	Bandwidth limit increment, as a percentage of the QoS factor for the zone	CHG ZQRT
QoS Response Time Interval (ZQRTI)	Time (in minutes) between bandwidth limit increments	CHG ZQRTI
Warning Alarm Threshold (ZQWAT)	A QoS value, which is lower than this value, but higher than the Critical (Unacceptable) Alarm Threshold, triggers a Major Alarm.	CHG ZQWAT
Critical Alarm Threshold (ZQUAT)	A QoS value, which is lower than this value, triggers an Unacceptable (Critical) Alarm.	CHG ZQUAT
R Alarm Coefficient (CR)	The R (Cr) coefficient is used to calculate the QoS value for the zone.	CHG CR
Packet Loss Alarm Coefficient (CPL)	The Packet Loss (Cpl) coefficient is used to calculate the QoS value for the zone.	CHG CPL
Delay Alarm Coefficient (CD)	The Delay (Cd) coefficient is used to calculate the QoS value for the zone.	CHG CD
Jitter Alarm Coefficient (CJ)	The Jitter (Cj) coefficient is used to calculate the QoS value for the zone.	CHG CJ
Coefficient of QoS (CQoS)	The Coefficient of QoS (CQoS) is used to calculate the overall QoS value for the zone.	CHG CQOS
Recent Validity Time Interval (CACVT)	Amount of time (in hours) for zone-to-zone record validity. Once this interval expires, records for unused zones are purged from the tables.	CHG CACVT

Configuration using Command Line Interface

You can also configure the Adaptive Network Bandwidth Management feature using LD 117.

LD 117 Configure Adaptive Network Bandwidth Management.

Command	Description
CHG CACVT <Zone> <Interval>	
	Configure the zone-to-zone record validity time interval, where: • Zone = 1-255 • Interval = 1-(48)-255

Command	Description
CHG CD <Zone> <Cd>	Change the Cd coefficient in the formula that determines how quickly an alarm reduces the Sliding Maximum bandwidth for the identified zone, where: • Zone = 1-255 • Cd = Cd coefficient = 1-(50)-100
CHG CPL <Zone> <Cpl>	Change the Cpl coefficient in the formula that determines how quickly an alarm reduces the Sliding Maximum bandwidth for the identified zone, where: • Zone = 1-255 • Cpl = Cpl coefficient = 1-(50)-100
CHG CJ <Zone> <Jitter>	Change the Cj coefficient in the formula that determines how quickly an alarm reduces the Sliding Maximum bandwidth for the identified zone, where: • Zone = 1-255 • Jitter = Jitter coefficient = 1-(50)-100
CHG CQOS <Zone> <QoS>	Change the QoS coefficient in the formula that determines how quickly an alarm reduces the Sliding Maximum bandwidth for the identified zone, where: • Zone = 1-255 • QoS = QoS coefficient = 1-(50)-100
CHG CR <Zone> <Cr>	Change the Cr coefficient in the formula that determines how quickly an alarm reduces the Sliding Maximum bandwidth for the identified zone, where: • Zone = 1-255 • Cr = Cr coefficient = 1-(50)-100
CHG ZONE <zoneNumber> <intraZoneBandwidth> <intraZoneStrategy> <interZoneBandwidth> <interZoneStrategy> [<zoneIntent> <zoneResourceType>]	

Command	Description
	Change the parameters of an existing zone, where: • zoneNumber = 1-255 • intraZoneBandwidth = 1000000 (Mbit/s) • intraZoneStrategy = intrazone preferred strategy — Best Quality = BQ — Best Bandwidth = BB • interZoneBandwidth = 100000 (Mbit/s) • interZoneStrategy = intrazone preferred strategy — Best Quality = BQ — Best Bandwidth = BB • zoneIntent = type of zone, where: — MO = Main office zone — BMG = Branch Media Gateway (branch office) zone — VTRK = Virtual Trunk zone • zoneResourceType = resource intrazone preferred strategy — shared DSP channels (default) = shared — private DSP channels = private ***Note:*** In CS 1000 Release 4.5, the zones that were described with BMG designator stay with BMG one, all the other zones are provided with MO designator. It is possible to update ZoneIntent using the CHG ZONE command.
CHG ZQRT <Zone> <Incr>	Change ZQRT, which is Response time increase by percentage. It is used to determine the increase to the Sliding Maximum for the identified zone, where: • Zone = 1-255 • Incr = increase value in percentage = 1-(10)-100
CHG ZQRTI <Zone> <Interval>	

Command	Description
	Change the QoS Response Time Interval while alarms are not coming, in order to increase the Sliding Maximum for the identified zone, where: • Zone = 1-255 • Interval = interval in minutes = 1-(5)-120
CHG ZQUAT <Zone> <Thres>	Change the QoS Unacceptable Alarm Threshold value for the identified zone, where: • Zone# = 1-255 • Thres = threshold value = 1-(75)-99 ***Note:*** When the zone-to-zone QoS value drops below the threshold value, the alarm is presented. This value must be below the value of ZQWAT.
CHG ZQWAT <Zone> <Thres>	Change the QoS Warning Alarm Threshold value for the identified zone, where: • Zone = 1-255 • Thres = threshold value = 1-(85)-99 ***Note:*** When the zone-to-zone QoS value drops below the threshold value, the alarm is presented. The value for ZQWAT must be higher than the value of ZQUAT.
CHG ZQNL <Zonenumber> <level>	Change the Notification Level for the specified zone, where: • Zone = 1-255 • Level = 0-(2)-4, where: — Level 0 = All voice quality alarms are suppressed. — Level 1 = All zone-based Unacceptable alarms. — Level 2 = Allow all level 1 alarms PLUS zone-based Warning alarms. — Level 3 = Allow all level 1 and 2 alarms PLUS per-call Unacceptable alarms.

Command	Description
	— Level 4 = Allow all level 1, 2, and 3 alarms PLUS per-call Warning alarms
NEW ZONE <zoneNumber> [<intraZoneBandwidth> <intraZoneStrategy> <interZoneBandwidth> <interZoneStrategy> <zoneIntent> <zoneResourceType>]	• zoneNumber = 1-255 • intraZoneBandwidth = 1000000 (Mbit/s) • intraZoneStrategy = BQ (Best Quality) • interZoneBandwidth = 1000000 (Mbit/s) • interZoneStrategy = intrazone preferred strategy — Best Quality = BQ — Best Bandwidth = BB • zoneIntent = type of zone, where: — MO = Main office zone — BMG = Branch Media Gateway (branch office) zone — VTRK = Virtual Trunk zone • zoneResourceType = resource intrazone preferred strategy — shared DSP channels (default) = shared — private DSP channels = private
DIS ZCAC <Zone>	Disables the Call Admission Control (CAC) feature for the specified zone, where: • Zone = 1-255 ***Note:*** Disables the feature on a zone-by-zone basis.

Command	Description
ENL ZCAC <Zone>	Enables the Call Admission Control (CAC) feature for the specified zone, where: • Zone = 1-255 ***Note:*** Enables the feature on a zone-by-zone basis.

Maintenance commands

The Adaptive Network Bandwidth Management feature can be maintained using Element Manager or LD 117.

Maintenance using Element Manager

The CAC parameters, intrazone statistics, and interzone statistics for one of more zones are available in Element Manager from the Zones web page, shown in Figure 50 "Zones web page" (page 122). To view the intrazone or interzone statistics, use Procedure 1 "Printing intrazone and interzone statistics for a zone" (page 124). To display the CAC parameters, follow the steps in Procedure 2 "Displaying CAC parameters for one or more zones" (page 142).

Procedure 2
Displaying CAC parameters for one or more zones

Step Action

1 In the EM Navigator select **System > IP Network > Zones**.

The **Zones** web page opens (see Figure 50 "Zones web page" (page 122)).

2 Click **Maintenance Commands for Zones (LD 117)**.

The **Maintenance Commands for Zones** web page opens, as shown in Figure 51 "Maintenance Commands for Zones web page" (page 125). This page lists all the configured zones and their intrazone statistics by default.

3 Select **Print Adaptive Network Bandwidth Management and CAC Parameters (PRT ZCAC)** from the **Action** drop-down list.

4 Select a zone from the **Zone Number** drop-down list, by doing one of the following:

- Select **ALL** to print statistics for all zones.

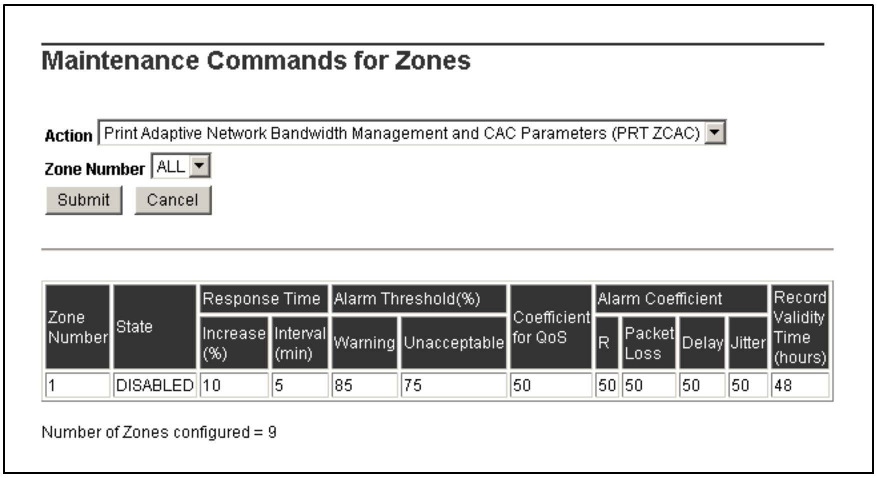
- Select a specific zone number to display statistics for a specific zone.

5 Click **Submit**.

The **Maintenance Commands for Zones** web page reopens, displaying the statistics for the specified zone or zones. A blank field indicates that the statistic is either not available or not applicable to that zone.

Figure 59 "Element Manager — CAC parameters" (page 143) shows an example of the CAC parameters for sample Zone 1.

Figure 59
Element Manager CAC parameters



Maintenance Commands for Zones

Action: Print Adaptive Network Bandwidth Management and CAC Parameters (PRT ZCAC)

Zone Number: ALL

Submit Cancel

Zone Number	State	Response Time		Alarm Threshold(%)		Coefficient for QoS	Alarm Coefficient				Record Validity Time (hours)
		Increase (%)	Interval (min)	Warning	Unacceptable		R	Packet Loss	Delay	Jitter	
1	DISABLED	10	5	85	75	50	50	50	50	50	48

Number of Zones configured = 9

—End—

Maintenance using LD 117

The same information can be displayed using commands in LD 117.

LD 117 Display Adaptive Network Bandwidth Management information

Command	Description
CLR CACR <Near Zone> [<Near VPNI>] [<Far Zone>] [{<Far VPNI>]	
	Clear zone-to-zone record for near (VPNI-Zone) for far (VPNI-Zone), where: • Near Zone = 1-255 • Near VPNI = 1-16383 • Far Zone = 1-255 • Far VPNI = 1-16383

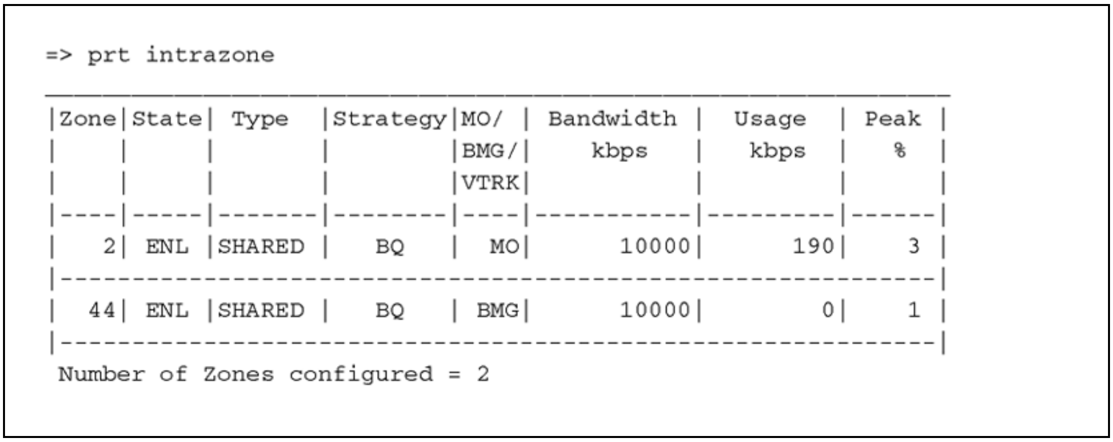
Command	Description
PRT INTRAZONE [<zone>]	
	Print intrazone statistics for the identified zones, where: • zone = ALL or 1-255 The output of this command displays the following information: • Zone • State = ENL/DIS • Type = PRIVATE/SHARED • Strategy = BB/BQ • MO/BMG/VTRK = ZoneIntent • Bandwidth = Kbps • Usage = Kbps • Peak = % Figure 60 "Sample output for PRT INTRAZONE command" (page 146) shows an example of the output for this command.
PRT INTERZONE [<nearZone>] [<nearVPNI>] [<farZone>] [<farVPNI>]	
	Print interzone statistics for the specific VPNI zone; where: • nearZone = ALL or 1-255 • nearVPNI = 1-16383 • farZone = 1-255 • farVPNI = 1-16383 The output of this command displays the following information: • Near end Zone • Near end VPNI • Far end Zone • Far end VPNI • State = ENL/DIS

Command	Description
	• Type = PRIVATE/SHARED • Strategy = BB/BQ • MO/BMG/VTRK = Zone Intent • QoS factor = % • Bandwidth configured = Kbps • Sliding max = Kbps • Usage = Kbps • Peak = % • Call = Cph • Alarm = Aph The report rows are grouped as: • First row = summary bandwidth usage per near zone • Next rows = bandwidth usage per near (VPNI- Zone) and far (VPNI - Zone) Figure 61 "Sample output for PRT INTERZONE command" (page 147) shows an example of the output for this command.
PRT ZCAC {<zone>}	Print CAC parameters for the specified zone, or for all zones, where: • zone = ALL or 1-255 The output of this command displays the following information: • Local ZONE = 1-255 • State = ENL/DIS • CR = 1-100 • CPL = 1-100 • CD = 1-100 • CJ = 1-100 • CQOS = 1-100

Command	Description
	• ZQRT = 1-100 • ZQRTI = 10-120 • ZQUAT = 1-99 • ZQWAT =1-99 • CACVT = 1-255

Sample outputs for PRT commands Figure 60 "Sample output for PRT INTRAZONE command" (page 146) shows an example of the output of the PRT INTRAZONE command. Figure 61 "Sample output for PRT INTERZONE command" (page 147) shows an example of the output of the PRT INTERZONE command.

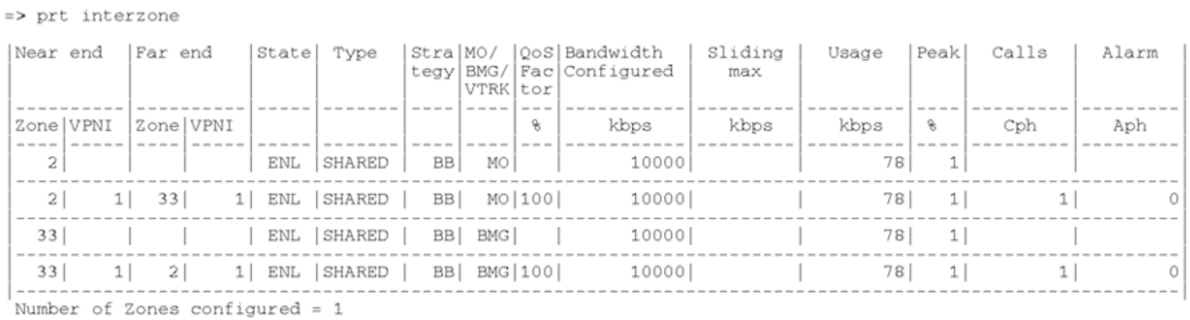
Figure 60
Sample output for PRT INTRAZONE command



```
=> prt intrazone
_______________________________________________________________
|Zone|State| Type  |Strategy|MO/ | Bandwidth |  Usage  | Peak |
|    |     |       |        |BMG/|   kbps    |  kbps   |  %   |
|    |     |       |        |VTRK|           |         |      |
|----|-----|-------|--------|----|-----------|---------|------|
|   2| ENL |SHARED |   BQ   |  MO|      10000|      190|   3  |
|-------------------------------------------------------------|
|  44| ENL |SHARED |   BQ   | BMG|      10000|        0|   1  |
|-------------------------------------------------------------|
 Number of Zones configured = 2
```

Conditional color code:
IP Peer only - blue
Branch office only - red
SRG only - magenta
IP Peer and Branch - cyan
IP Peer and SRG - yellow
Branch and SRG - green
None

Figure 61
Sample output for PRT INTERZONE command



```
=> prt interzone

|Near end  |Far end   |State| Type   |Stra|MO/ |QoS|Bandwidth  | Sliding  |  Usage   |Peak|  Calls   |  Alarm   |
|          |          |     |        |tegy|BMG/|Fac|Configured |   max    |          |    |          |          |
|          |          |     |        |    |VTRK|tor|           |          |          |    |          |          |
| ---------- ---------- ----- -------- ---- ---- --- ----------- ---------- ---------- ---- ---------- ---------- |
|Zone|VPNI |Zone|VPNI |     |        |    |    | % |   kbps    |   kbps   |   kbps   | %  |   Cph    |   Aph    |
| ---- ----- ---- ----- ----- -------- ---- ---- --- ----------- ---------- ---------- ---- ---------- ---------- |
|   2|     |    |     | ENL |SHARED  |  BB|  MO|   |     10000|           |        78|   1|          |          |
| -------------------------------------------------------------------------------------------------------------- |
|   2|    1|  33|    1| ENL |SHARED  |  BB|  MO|100|     10000|           |        78|   1|         1|         0|
| -------------------------------------------------------------------------------------------------------------- |
|  33|     |    |     | ENL |SHARED  |  BB| BMG|   |     10000|           |        78|   1|          |          |
| -------------------------------------------------------------------------------------------------------------- |
|  33|    1|   2|    1| ENL |SHARED  |  BB| BMG|100|     10000|           |        78|   1|         1|         0|
| -------------------------------------------------------------------------------------------------------------- |
 Number of Zones configured = 1
```

Note: The Far end and VPNI fields are displayed only when Adaptive Bandwidth Management is enabled in LD 117.

Features

Contents

This section contains information on the following topics:

Tone handling

Progress tones

The IP Phone or Gateway can generate call-progress tones locally. IP Peer Networking supports both in-band and out-of-band generated tones. For example, simple calls between IP Phones rely exclusively on out-of-band locally generated tones. A call from an IP Phone to an analog Gateway (or to an ISDN Gateway that terminates on an analog line) can rely exclusively on in-band tones. The state of the terminating side is not always known by the originating end through the H.323 protocol or SIP. Therefore, some scenarios require generating in-band tones from the terminating side.

Dial tone is always the responsibility of the originating side. The call is not setup with the far end as long as the digits are gathered for en-bloc transmission, or for overlap signaling until the provisioned minimum number of required digits is met on the Call Server. Other tones are provided by the originating side when the call cannot proceed to the far end.

For calls that terminate within a private network of CS 1000 systems, ringback tone is provided locally at the originating Call Server. This is based on the tone definition within that Call Server. Calls terminating on analog trunk gateways relay the tone generated from the PSTN through to the originator of the call.

Call modification scenarios, after a call has been answered, result in the provision of in-band tones. In this case, the generated tones are determined by the flexible tone configuration at that Call Server (that is, where the modification occurred).

In-band tones are generated by connecting a Tone circuit to a DSP channel so that the tone samples can be transported across the IP network within standard RTP streams.

For call center limitations on tone handling, see the "Limitations" (page 169).

End-to-end DTMF signaling

Dual Tone Multifrequency (DTMF) signaling represents the pressing of dialpad keys (0-9, *, #) on a telephone during a call. IP Peer Networking supports the sending and receiving of DTMF signaling during speech.

DTMF signaling can be received from the following:

- analog (500/2500-type) telephones
- digital telephones
- IP Phones
- Virtual Trunk (SIP/H.323 trunks)
- analog trunks
- PRI trunks

Standard SIP and H.323 protocols are used to transmit DTMF tones.

> ***Note:*** IP Peer Networking does not support long DTMF tones over Virtual Trunks. Long-digit duration is not supported.

Tone handling methods

DTMF tones must be transmitted using out-of-band signaling, because sources of delay and distortion caused by IP media streams can cause invalid tone detection when transmitted in-band.

- The out-of-band method uses H.245 channel signaling messages to represent the DTMF tones for H.323.
- The out-of-band method uses INFO methods to represent the DTMF tones for SIP.

Out-of-band signaling

Out-of-band DTMF tones are generally used for Virtual Trunks. The DTMF tones are sent as messages over the signaling channels. The messages are then converted to tones on the receiving side. This is a reliable way of sending DTMF tones over the Virtual Trunk.

SIP End-to-end DTMF signaling is carried out-of-band by the SIP INFO message. The message does not include information about the duration of DTMF tones, and, as a result, long DTMF tones are not supported.

The INFO format is the same as MCS 5100 implementation. However, third-party gateways may use a different INFO format or even a different method to implement the out-of-band DTMF, which might lead to an interoperability issue. For more information, refer to RFC 2976 – The SIP INFO Method.

H.323 Out-of-band DTMF tones are transmitted using H.245 *UserInputIndication* messages. The content of each message represents the key that generated the tone. The message can represent the key value using a string indication, a signal indication, or both. If the signal indication is used, the message can also include a parameter to represent the tone method duration (that is, how long the key was pressed).

The endpoints negotiate which method is used. This negotiation occurs during H.323 call setup signaling.

On receipt of a *UserInputIndication* message, the receiving H.323 Signaling proxy signals the appropriate entity to generate the corresponding tone. This depends on whether the call involves a circuit-switched party or an IP party. DTMF Tone Detection is a configurable codec parameter.

> ***Note:*** In-band DTMF tones that originate from an analog (500/2500-type) telephone or incoming trunk are filtered out of the media stream by the DSP of the Voice Gateway Media Card. This is so

that double detection of the DTMF digits does not occur. This causes additional delay in the speech path due to the buffering required to ensure that no DTMF tones get through the filter.

In-band signaling

In-band DTMF tones are sent as RTP packets over the RTP channels. This method of transporting DTMF tones is inherently unreliable as RTP packets can be lost over the network. However, this method is quite reliable if a G.711 codec is used for the transmission.

For CS 1000 systems, the in-band DTMF tones can be sent only from an analog (500/2500-type) telephone with tone detection turned off for the Voice Gateway Media Card.

IP Phone End-to-End Signaling (EES)

An IP Phone uses UNISTIM messages to signal digits. These messages are received by the telephone’s Terminal Proxy Server (TPS), which translates the messages into SSD format for existing call processing.

IP Phones EES to H.323 trunks

On receipt of a message that represents a key press on an IP Phone, the Call Server relays it to the H.323 Signaling Proxy. The H.323 Proxy generates the appropriate H.245 *UserInputIndication* message.

Circuit-switched device DTMF and EES

Circuit-switched devices can transmit DTMF tone using the circuit-switched switching fabric or using SSD messages in the case of EES. When a circuit-switched device connects to a remote party over an H.323 trunk, the circuit-switched gateway (DSP) detects the DTMF tone and informs the Call Server. The Call Server signals the H.323 Signaling Proxy to generate an H.245 *UserInputIndication* message to represent the tone. When a digital telephone is operating the EES feature, the Call Server receives the input message and behaves as described below.

DTMF signaling for a circuit-switched trunk and analog (500/2500-type) telephones using H.323 trunks

During call setup, a Digitone Receiver (DTR) is connected to the circuit-switched trunk or analog (500/2500-type) telephone if DTMF is used for dialing. Digits detected for call setup are handled the same way as traditional call processing.

After a call has been established, circuit-switched trunks (for example, PRI trunks) or 2500 lines can carry DTMF tones in-band. When a circuit-switched trunk or analog (500/2500-type) telephone is connected to an H.323 trunk, tones are passed through the circuit-switched switching fabric to the circuit-switched gateway (DSP). The DSP detects the DTMF tone and informs the Call Server. The Call Server signals the H.323 Signaling Proxy to generate an H.245 *UserInputIndication* message to represent the tone.

DTMF out-of-band signals from H.323 trunk

For calls incoming from an H.323 trunk, DTMF signals are indicated using the H.245 *UserInputIndication* message.

Calls from H.323 trunks to circuit-switched trunks/analog (500/2500-type) telephones/digital telephones

On receipt of an H.245 *UserInputIndication* message, the H.323 Proxy signals the circuit-switched gateway (DSP) that supports the circuit-switched call. This is to generate the appropriate DTMF tone through the circuit-switched switching fabric to the terminating circuit-switched device.

> ***Note:*** Out-of-band DTMF signals received when a Virtual Trunk is connected to an IP Phone are ignored and not sent to the IP Phones.

Tandem H.323 trunks to H.323 trunks

On receipt of an H.245 *UserInputIndication* message on a given signaling proxy, the proxy transmits an appropriate *UserInputIndication* message on the connected outgoing H.323 signaling channel.

DTMF out-of-band signals from SIP trunk

For calls incoming from a SIP trunk, DTMF signals are indicated using the SIP INFO message.

Calls from SIP trunks to circuit-switched trunks/analog (500/2500-type) telephones/digital telephones

On receipt of a SIP INFO message on a given SIP Trunk Gateway, the SIP Trunk Gateway transmits an appropriate message to the Call Server. The Call Server then relays the message to the other SIP Trunk Gateway, which then sends out a SIP INFO message.

This generates the appropriate DTMF tone through the circuit-switched switching fabric to the terminating circuit-switched device.

> ***Note:*** Out-of-band DTMF signals received when a Virtual Trunk is connected to an IP Phone are ignored and not sent to the IP Phones.

Tandem SIP trunks to SIP trunks

On receipt of a SIP INFO message on a given signaling SIP Trunk Gateway, the SIP Trunk Gateway transmits an appropriate SIP INFO message on the connected outgoing SIP signaling channel.

> ***Note:*** RFC2833 allows interoperability with other SIP products that do not support out-of-band DTMF tones.

Fax calls

SIP

T.38 UDP fax is supported. The switchover procedure in T.38 ANNEX D (D.2.2.4) is used to establish a fax channel.

A SIP INVITE is made to the called party requesting a voice connection using the basic call setup flow. A voice connection is then established. Upon the detection of the fax tone (V.21) at the terminating end, the voice channel is replaced by a fax channel using the offer/answer SDP exchange.

H.323

IP Peer Networking supports the voice-to-fax switchover protocol for T.38 fax, by using the mode select signaling in H.323.

First, a voice call is established. When the DSP detects the fax tone, H.245 signaling is exchanged to request the far end node to change from voice mode to T.38 mode. The existing voice channels are closed and new channels for T.38 are opened. The fax call then proceeds.

The CS 1000 systems comply with H.323 version 3.0 with the H.323 version 4.0 extensions necessary for voice-to-fax switchover. This version standardizes the procedures in switching from voice mode to fax mode. Some third-party H.323 gateways can use different implementations of protocols to switch from voice to fax. Using a third-party gateway requires fax interoperability testing of the system. The end result can be that fax is not supported, due to the complexity of the H.323 protocol and other factors. Check with your Nortel sales representative for approved third-party gateways.

Nortel does not recommend using a modem on the CS 1000 network, due to the variety of modems available and the issues of packet loss and delay. For more information about fax support and limitations, see *IP Trunk: Description, Installation, and Operation* (NN43001-563).

Reliability and redundancy

CS 1000 systems provide levels of redundancy to ensure that telephony services can withstand single hardware, software, and network failures. Table 10 "Reliability and redundancy features by system type" (page 155) shows each reliability and redundancy feature and the systems that support the feature. The reliability and redundancy features include:

- "Alternate Call Server" (page 155)

- "Signaling Server software redundancy" (page 157) (including H.323/SIP Trunk Gateway and IP Phone software)
- "H.323 Gateway software — trunk route redundancy" (page 158) (H.323 Gateway interface to Gatekeeper redundancy [Failsafe Gatekeeper])
- "SIP Trunk Gateway software — trunk route redundancy" (page 158)
- "NRS redundancy" (page 160)
- "Campus-distributed Media Gateway in survival mode" (page 164)
- "CS 1000M Large System CPU redundancy" (page 165)

Table 10 "Reliability and redundancy features by system type" (page 155) shows the features and the systems that support the feature.

Table 10
Reliability and redundancy features by system type

Reliability and Redundancy Features		**CS 1000M Large Systems**	
	CS 1000E	**CS 1000M Single Group**	**CS 1000M Multi Group**
Alternate Call Server	X		
Signaling Server software redundancy	X	X	X
NRS redundancy	X	X	X
SIP Trunk Gateway	X	X	X
H.323 Gateway	X	X	X
Campus distributed Media Gateway in survival mode	X		
CPU redundancy		X	X
Survivable IP Expansion (SIPE)	X		

Note 1: For CS 1000E redundancy, refer to *Communication Server 1000E: Planning and Engineering (NN43041-220)*.

Note 2: For Geographic Redundancy, refer to *System Redundancy Fundamentals (NN43001-507)*.

Alternate Call Server

All Media Gateways have a full set of call-processing software components and maintain a configuration database that is periodically synchronized with the primary Call Server.

During normal operation, the processor in the Media Gateway handles low-level control of the interface cards in the gateway slots and communicates with the Call Server for feature operation. If the Media Gateway processor loses communication with the Call Server due to Call Server or IP network component failure (for example, cabling and

L2 switch), one Media Gateway, configured as the Alternate Call Server, assumes Call Server responsibilities. The Signaling Server registers with that Alternate Call Server. Other Media Gateways can access only local Gateway hardware and local non-IP Phones, and are not under the control of the Alternate Call Server.

The Alternate Call Server IP address must be in the same ELAN subnet as the Primary Call Server IP address.

The Alternate Call Server is applicable only to the CS 1000E system and CS 1000M Small Systems.

As an example,Figure 62 "Example Normal mode of operation for a CS 1000E system" (page 156) shows the normal mode of operation for a CS 1000E system.

Figure 62
Example Normal mode of operation for a CS 1000E system

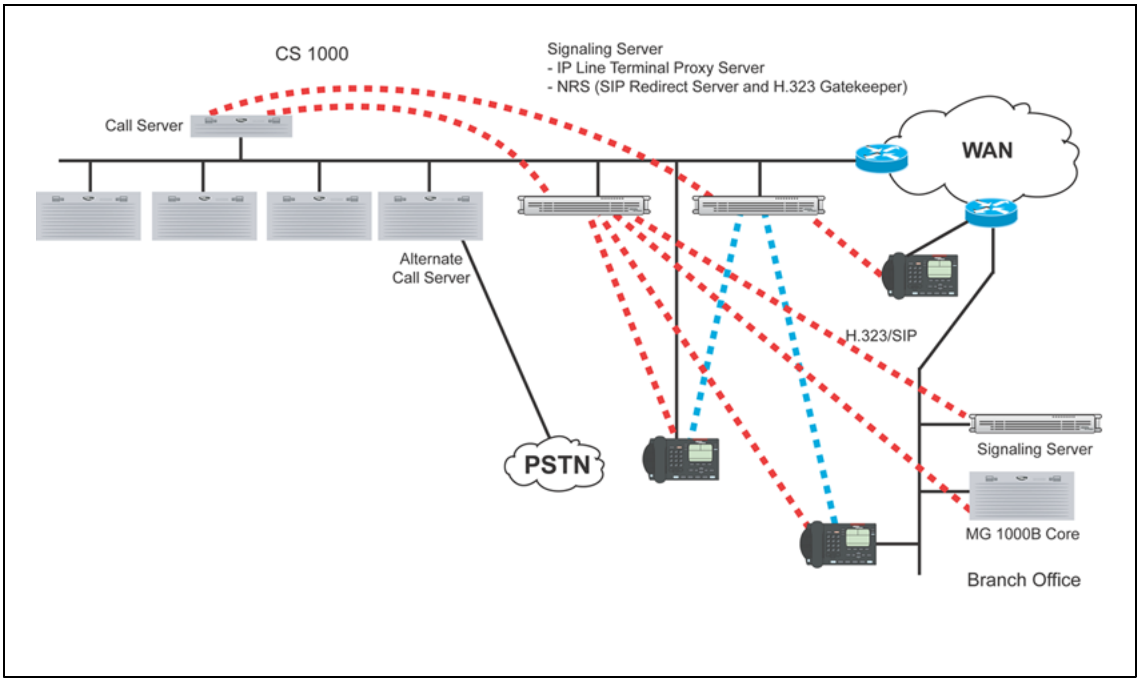
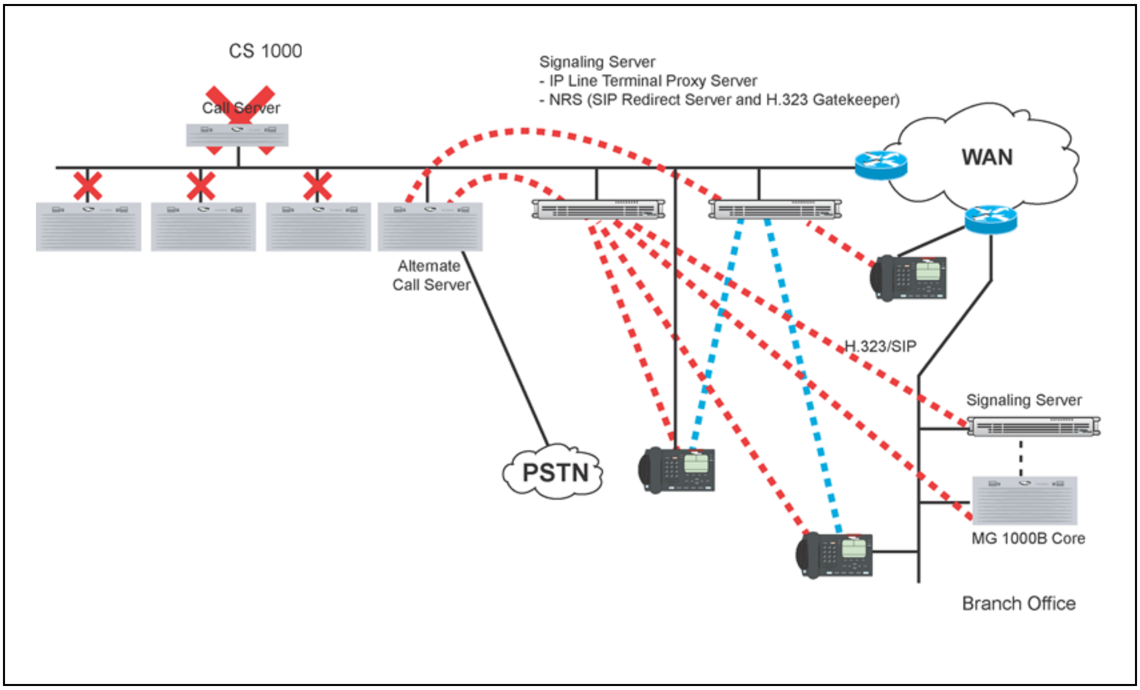


Figure 63 "Call Server failure and redundancy in a CS 1000E system" (page 157) illustrates what occurs when the Call Server in a CS 1000E system fails:

1. The Call Server database periodically synchronizes at the Alternate Call Server.
2. The Primary Call Server fails.
3. The Alternate Call Server assumes the role of Primary Call Server for IP Phones.
4. The Signaling Server registers at the Alternate Call Server.

5. Operation resumes with all Media Gateways, but the Signaling Server registers only with the Alternate Call Server.

Figure 63
Call Server failure and redundancy in a CS 1000E system



Signaling Server software redundancy

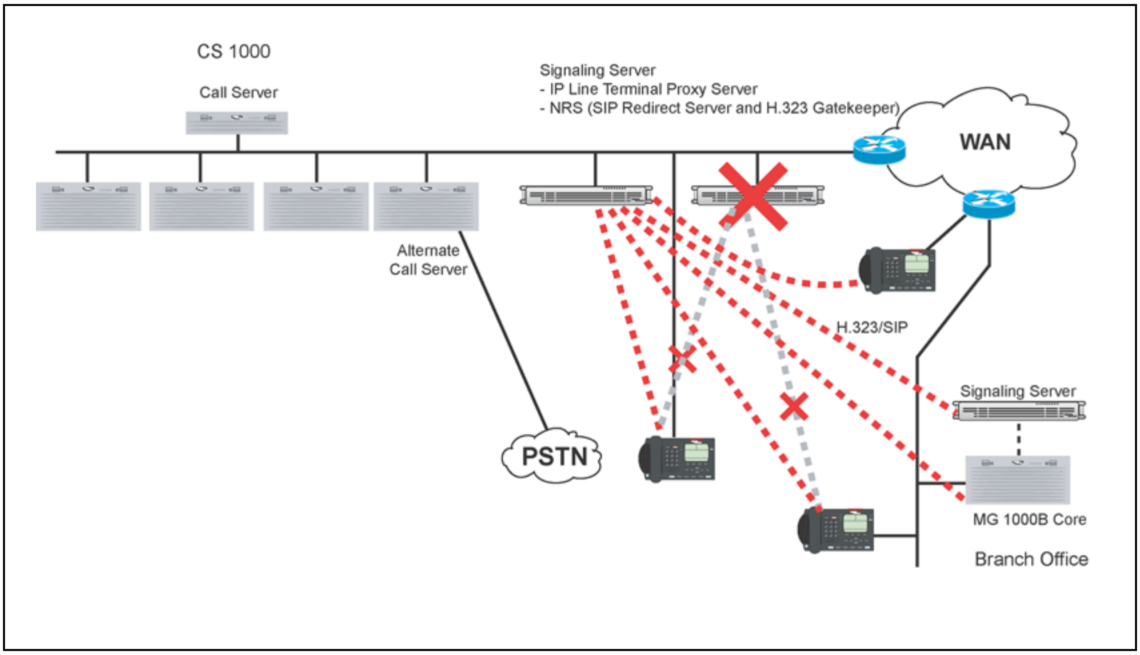
Signaling Server redundancy is provided on a load-sharing basis for the TPS. The Follower Signaling Server is the platform for the SIP/H.323 Gateway software if the Leader Signaling Server fails. The NRS (Primary, Alternate, or Failsafe) cannot reside on a Follower Signaling Server. It must reside on a Leader Signaling Server.

As an example,Figure 64 "Signaling Server redundancy in a CS 1000E system" (page 158) shows Signaling Server redundancy in a CS 1000E system. In the example, the following occurs:

1. The IP Phones are distributed between the two Signaling Servers. The SIP/H.323 Gateway software runs on the Primary Signaling Server.
2. The Primary Signaling Server fails.
3. The Alternate Signaling Server assumes the Connection Server IP address, if necessary.
4. The IP Phones Time-to-Live time-out causes the IP Phones to reset and register to the Alternate Signaling Server. Active calls are dropped.
5. The Alternate Signaling Server assumes responsibility for the SIP/H.323 Gateway software.

6. Operation resumes.

Figure 64
Signaling Server redundancy in a CS 1000E system



H.323 Gateway software trunk route redundancy

The H.323 Gateway software runs on the Node Master. The Signaling Server is normally configured as the Leader. If the Primary (Leader) Signaling Server fails, an Alternate (Follower) Signaling Server can take over the Node IP address. The Gateway software then runs on the Signaling Server with the Node IP address.

Existing calls are kept when the Primary Signaling Server fails. This situation applies to IP Phones that are not registered with the Primary Signaling Server, and for all circuit-switched telephones. IP Phones that are registered with the Primary Signaling Server restart after the Time-to-Live time-out, so active calls on those telephones are lost.

SIP Trunk Gateway software trunk route redundancy

Each Call Server can have one or more SIP nodes; however, at any time each node has only one active gateway. A separate Signaling Server can be configured to run the SIP Trunk Gateway application as a backup (or alternate SIP Trunk Gateway). SIP Trunk Gateway redundancy is similar to the H.323 Gateway redundancy implementation. That is, the Leader and Follower Signaling Servers are configured in the same node. If the Leader Signaling Server fails, the Follower Signaling Server with the Alternate SIP Trunk Gateway becomes the master and takes over the node IP.

All active calls remain active during switchover; however, a near-end call is completely released using Scan and Signal Distributor (SSD) messages when the near-end party hangs up the call.

If the Leader (Primary) SIP Trunk Gateway comes back up during active calls, the following occurs:

- The busy channels stay busy in the Alternate SIP Trunk Gateway.
- The idle channels register with the Primary SIP Trunk Gateway.
- The near-end calls are released from the Alternate SIP Trunk Gateway when the near-end party hangs up. The SIP Virtual Trunk channels then register with the Primary SIP Trunk Gateway.

Each SIP Trunk Gateway occupies one Virtual Trunk route. To have SIP and H.323 Virtual Trunks co-residing on the same Signaling Server platform, the Virtual Trunks must be configured in separate routes, but must use the same IP D-channel ID.

SIP Proxy Trunk Gateway software trunk route redundancy

The following scenario applies to the SIP Proxy Trunk gateway.

When active calls are switched from a follower Signaling Server to a leader Signaling Server and vice versa:

1. Since this is not a hot standby redundancy and non-loadsharing implementation, there will be a complete loss of service for virtual trunking for 2 to 30 seconds when switching from the leader to follower Signaling Server. (The loss of service will be about 10 seconds if the leader goes down or if the TLAN goes down and 2 minutes if the ELAN goes down).
2. When a VTRK switch over occurs, all of the idle VTRKs will register to the new master and active VTRKs will be kept on the old master.
3. When a call over a Signaling Server is released from the far end, the follower is not the master anymore and the near end VTRK is not released until the near end hangs up. If for some reason the near end fails to hang up the VTRK stays busy.
4. When a call over a Signaling Server is released from the near end, the follower is not the master anymore and the far end VTRK is not released until the far end hangs up. If for some reason the far end fails to hang up the VTRK stays busy.

NRS redundancy

The NRS provides address translation services for all endpoints in the network zone; therefore, redundancy is important. If an endpoint cannot reach an NRS over the network for address translation, calls cannot be placed. Nortel recommends that a backup or Alternate NRS be installed and configured on the network.

The CS 1000 networks have at least one NRS to provide network numbering plan management for private and public numbers. An optionally redundant NRS can be installed in the network. The Alternate NRS periodically synchronizes its database with the Primary NRS.

Primary, Alternate, and Failsafe NRS databases are supported. The H.323 or SIP Trunk Gateway software attempts to recover system functionality if a failure occurs at the NRS. The two types of NRS redundancy are:

- Alternate NRS
- Failsafe NRS

The Alternate NRS is optional but recommended for all networks. The Failsafe NRS is also optional but is recommended for selected critical IP Peer H.323 and SIP Trunk Gateways.

Only one of the servers in the pair is active at one time — the other is on standby. A heartbeat mechanism between servers is implemented. When a failure of the heartbeat from the active server is detected, the standby server takes over. Another mechanism ensures that both servers have up-to-date configuration.

By optimizing timeout and threshold parameters used in retries of the heartbeat mechanism, ungraceful switchover trigger time is reduced to less than 15 seconds with CS Release 5.0. The optimization in the timing leads to a change in the INI policy. When the active core warm starts, the inactive core also reboots, so no swapping of the cores takes place.

For NRS/H.323 Gatekeeper redundancy, see below.

For NRS/SIP Redirect Server redundancy, see "NRS SIP Redirect Server redundancy" (page 163).

NRS H.323 Gatekeeper redundancy

Alternate H.323 Gatekeeper The H.323 Gateway software runs on the Signaling Server and communicates with both a Primary and Alternate (optional) H.323 Gatekeeper. If the Gateway software loses communication with its Primary H.323 Gatekeeper, it automatically registers at the Alternate H.323 Gatekeeper to resume operation.

To enable the Alternate H.323 Gatekeeper to provide H.323 Gatekeeper redundancy, the CS 1000 systems can accept a prioritized list of Alternate NRSs in the Gatekeeper Confirmation (GCF) and Registration Confirmation (RCF) messages returning from the Primary Gatekeeper at the Gatekeeper Discovery and Gatekeeper Registration times respectively.

> ***Note:*** The list of Alternate Gatekeepers in the registration confirmation message takes precedence over the list in the Gatekeeper confirmation message. At any time, if the system detects that it is not registered, or if the Gatekeeper does not respond (for example, because it receives an Unregister Request (URQ) message or because the Time-to-Live messages are not answered), it reattempts registration to its Primary Gatekeeper (the address that was returned by the GCF). The value of the Time-to-Live timer is determined by the Gatekeeper in the RCF, and obeyed by the endpoint. If the timer fails, the system sequentially attempts to register with the Alternate Gatekeepers until registration succeeds.

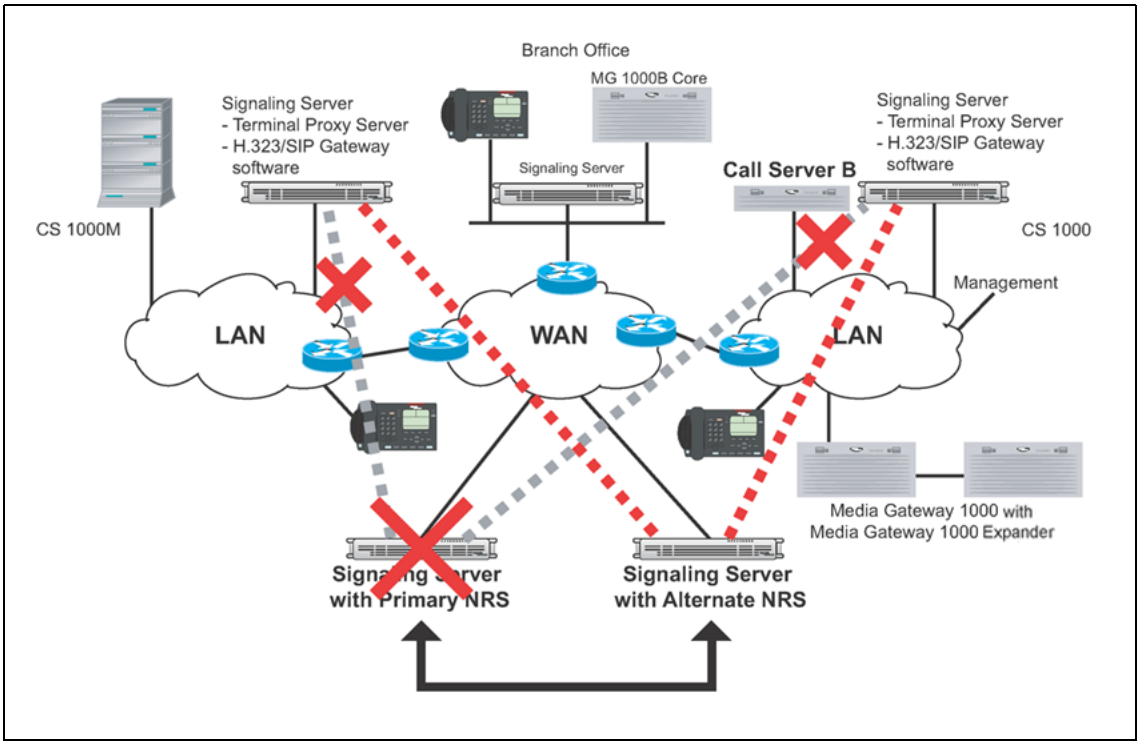
Polling and switchover A Time-to-Live timer is provided to ensure that if a Gatekeeper stops responding for a specified amount of time, the H.323 Gateway software registers at the Alternate Gatekeeper to resume operation. This ensures Gatekeeper redundancy across the network.

The Alternate Gatekeeper is inactive and in standby mode by default. It constantly polls the Primary Gatekeeper by sending Information Response Request (IRR) messages to the Primary Gatekeeper. The default for the poll interval is configured to approximately 30 seconds and can be configured through NRS Manager (see *Configuring system-wide settings*). The *endpointType.gatekeeper* field of the IRR message is configured to indicate that the IRR is coming from a Gatekeeper and not an endpoint. If the Primary Gatekeeper is currently in-service and accepting registrations, then it returns an *Information Request Negative Acknowledgement (INAK)* message with *nakReason* set to *notRegistered*.

Figure 65 "Primary NRS failure and redundancy" (page 162) shows the handling of the Gateway interface and the Alternate Gatekeeper in the event of Primary Gatekeeper failure:

1. The Alternate Gatekeeper periodically synchronizes with the Primary Gatekeeper.
2. The Primary Gatekeeper fails.
3. The Alternate Gatekeeper assumes the role of the Primary Gatekeeper and generates a Simple Network Management Protocol (SNMP) alarm.
4. The Gateways time out and register at the Alternate Gatekeeper.
5. The network calls resume.

Figure 65
Primary NRS failure and redundancy



In addition to Gatekeeper redundancy, the H.323 Gateway interfaces can withstand communication loss to both Gatekeepers by reverting to a locally cached copy of the Gateway addressing information. Since this cache is static until one of the Gatekeepers becomes accessible, it is intended only for a brief network outage.

Failsafe H.323 Gatekeeper For additional redundancy, provide a Failsafe Gatekeeper at each endpoint in the network.

When configuring the Gatekeeper, the administrator must configure whether the Gatekeeper is the Primary Gatekeeper (GKP) or the Alternate Gatekeeper (GKA). If the Gatekeeper is the Primary Gatekeeper, the administrator can statically configure the IP address of the GKA (if an Alternate Gatekeeper is used on the network). If the H.323 Proxy Server application on the Signaling Server cannot contact the Primary or Alternate Gatekeepers, it can fall back on its local Failsafe Gatekeeper. Failsafe Gatekeepers are used only by local Signaling Server components. Failsafe Gatekeepers reject all Registration, Admission, and Status signaling (RAS) messages received over the network from remote entities. The Failsafe Gatekeeper provides a Security Denied messages.

The Primary Gatekeeper returns the IP address of the Alternate Gatekeeper (if an Alternate Gatekeeper is configured) in the *alternateGatekeeper* field of GCF and RCF messages. The Alternate Gatekeeper returns the IP address of the Primary Gatekeeper in the *alternateGatekeeper* field of GCF and RCF messages.

> ***Note:*** If the endpoints are configured with the IP addresses of Primary and Alternate Signaling Servers, the IP addresses, which are returned in the GCF and RCF messages, take precedence over configured IP addresses.

NRS SIP Redirect Server redundancy

Alternate SIP Redirect Server Normally only the Primary SIP Redirect Server is the active SIP Redirect Server. The Primary SIP Redirect Server has the master database while the Alternate SIP Redirect Server and Failsafe SIP Redirect Server have a replica of the database.

If the master database is changed, the Primary SIP Redirect Server creates a publication for the replica. The replica database automatically synchronizes the database from the master.

> ***Note:*** A user can also force a manual database synchronization.

The database synchronization success and failure messages are logged in the RPT report log.

Polling and switchover A polling message is sent out between Primary and Alternate SIP Redirect Servers and between Primary and Failsafe SIP Redirect Servers.

If the Alternate SIP Redirect Server determines that the Primary SIP Redirect Server is unreachable, it automatically switches to become the active SIP Redirect Server and its database becomes the master database. At the same time, the Failsafe SIP Redirect Server also determines that Primary is no longer available and it automatically switches to contact the Alternate SIP Redirect Server. The replica database on the Failsafe synchronizes with the master database on the Alternate SIP Redirect Server, if required.

Once the Primary SIP Redirect Server become inactive, no configuration changes are allowed. Only maintenance operations can be performed.

Switch-over messages are logged in the RPT report log.

Failsafe SIP Redirect Server If the Failsafe SIP Redirect Server loses its connection with both the Primary and Alternate SIP Redirect Servers, then it becomes the active SIP Redirect Server. When a failsafe GW registers to itself it always sends the registration message over UDP.

Campus-distributed Media Gateway in survival mode

In addition to having an Alternate Call Server, you can have Survivable Media Gateways (each of the Media Gateways can be survivable).

The Media Gateway survival modes applies to the following systems:

- CS 1000E System
- CS 1000M System

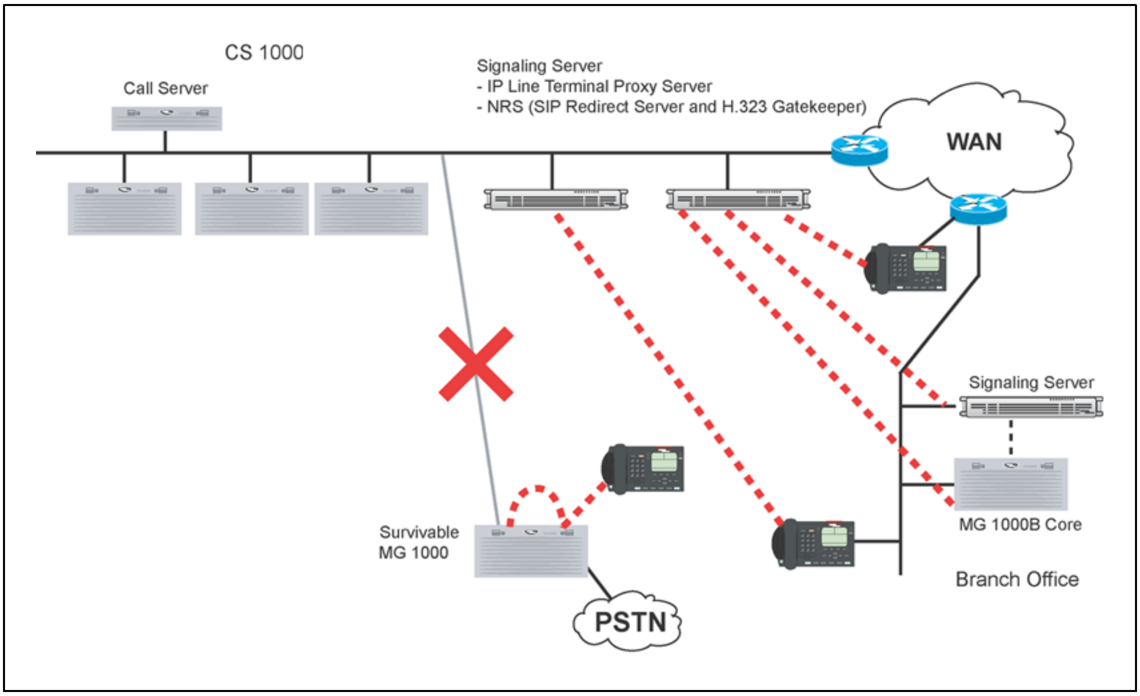
Media Gateways can be configured as survivable when distributed throughout a campus environment. In this case, basic telephony services are provided in the event of a network outage. Figure 66 "Network failure with Survivable Media Gateways" (page 165) illustrates how such an outage is handled.

The following list indicates the steps to a call in the survival mode scenario:

Step	Action
1	The Call Server database periodically synchronizes at the campus-distributed Media Gateway.
2	The Primary Call Server fails.
3	The campus-distributed Media Gateway assumes the role of the Primary Call Server for IP Phones.
4	The Signaling Server registers at the campus-distributed Media Gateway.
5	Operation resumes with the single Media Gateway.

—End—

Figure 66
Network failure with Survivable Media Gateways



> ***Note:*** To facilitate the survival mode operation below, the IP address configured in the IP Phones (for example, through DHCP) must be the Node IP address of the Voice Gateway Media Cards in the Survivable Media Gateway.

CS 1000M Large System CPU redundancy

The CS 1000M Large Systems have dual hot standby CPU redundancy to handle failure of the Call Server. IP Peer Networking supports the following Large System redundancy features:

- Health Monitoring
- Virtual Trunk redundancy
- Graceful switch-over
- Ungraceful switch-over

Health Monitoring

The health of the dual CPUs are monitored such that the active CPU switches over to the standby CPU when the standby CPU is healthier than the active CPU. The health of a CPU is calculated based on the conditions of various system components. For IP Peer Networking, the Signaling Server is one of the monitored components. If a CPU switch-over occurs, the Signaling Server registers with the new CPU.

The Signal Server uses the IP Line scheme for health monitoring. This scheme has a minimum threshold of two (that is, at least two IP Line connections) must exist before the health count is initiated. As a result, two Signaling Servers are required for health monitoring to work.

Table 11 "Health count" (page 166) shows the health count scheme.

Table 11
Health count

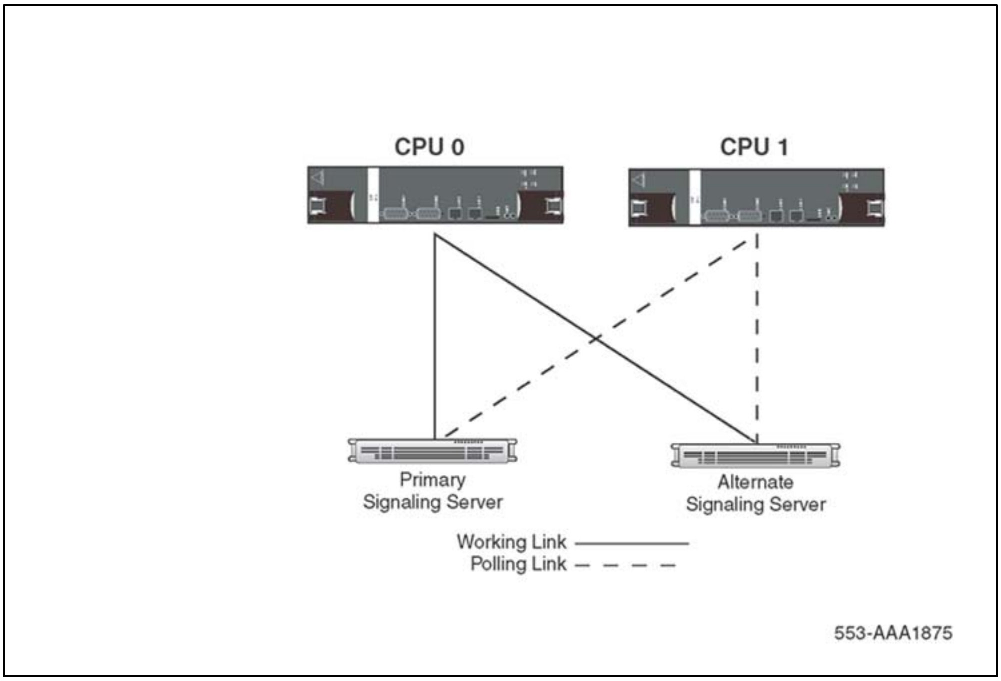
Number of cards	Health count
2 or 3 cards	1 health count
4 or 5 cards	2 health counts
6 or 7 cards	3 health counts
8 or 9 cards	4 health counts
...	...

Under normal operation, the following occurs:

- The primary Signaling Server works with the active CPU (CPU 0) over a working link and also keeps contact with the standby CPU (CPU 1) over a polling link.
- The alternate Signaling Server keeps contact with the active CPU (CPU 0) over a working link and the standby CPU (CPU 1) over a polling link.

Figure 67 "Health Monitoring" (page 167) illustrates health monitoring under normal operation.

Figure 67
Health Monitoring



When all the links are up and running there is no CPU switch-over. However, if the ELAN network interface in the active CPU (CPU 0) stops working, both Signaling Servers cannot communicate with the active CPU, and the health count on the active CPU is decreased. The health count of the standby CPU remains the same because both Signaling Servers can communicate with it.

Therefore, the standby CPU is healthier. A CPU switch-over takes place, and the standby CPU becomes the active CPU. The primary Signaling Server registers with the new active CPU.

Virtual Trunk redundancy

If the ELAN network interface on the Primary Signaling Server fails or the server itself fails, no CPU switch-over occurs, because both the active and the standby CPU lose contact with the Primary Signaling Server. As a result, they have the same health count.

The Virtual Trunk Redundancy mechanism is initiated. If a Virtual Trunk is unavailable, the call-processing software selects an alternate route. The alternate Signaling Server becomes the master and registers to the active CPU to resume the Virtual Trunk operation. The transient calls are dropped, while the established calls remain. The alternate Signaling Server becomes active in approximately 30 seconds, but calls cannot be initiated during that time.

Graceful switch-over

During a graceful switch-over, both established calls and transient calls survive the CPU switch-over. When the connection between the Signaling Servers and the active CPU goes down, a graceful switch-over occurs so that the Signaling Servers can register to the standby CPU that has become active. There is no impact to the calls; however, the report log file shows that graceful switch-over has taken place.

Ungraceful switch-over

During an ungraceful switch-over, the standby CPU sysloads and then everything returns to a normal state. For IP Peer Networking, the Signaling Server registers to the standby CPU. The report log file shows that ungraceful switch-over has taken place.

Least Cost Routing

IP Peer Networking supports the traditional methods of managing costs in a circuit-switched environment (for example, through BARS/NARS). See *Basic Network Features* (NN43001-579).

IP Peer Networking also supports a method to manage costs at the NRS. This is done in an IP environment using Least Cost Routing. With Least Cost Routing, you can assign a cost factor to the routes using NRS Manager. You can also use Least Cost Routing to identify the preferred SIP/H.323 Gateways for specific numbering plan entries. See *Adding a Routing Entry*.

Licenses

For each Virtual Trunk configuration, you must purchase a License. The number of trunks must match those that are enabled with the installation keycode.

The following packages are available for IP Peer Networking:

- H.323 Virtual Trunk (H323_VTRK) package 399
- SIP Gateway and Converged Desktop Package (SIP) package 406

The following Licenses are available for IP Peer Networking:

- SIP Access Port License
- H.323 Access Port License

For more information, refer to the following NTPs.

- *Communication Server 1000M and Meridian 1: Small System Installation and Commissioning* (NN43011-310)
- *Communication Server 1000M and Meridian 1: Large System Installation and Commissioning* (NN43021-310)

- *Communication Server 1000E: Installation and Commissioning* (NN43041-310)

Limitations

The NRS (Primary, Alternate, or Failsafe) cannot reside on an Alternate Signaling Server. It must reside on a Primary (Leader) Signaling Server.

Circuit capacity can provide a maximum of 60 simultaneous channels for tone generation and handling. Some queuing is provided when a channel becomes available. In order to alleviate the number of tone channels required for call center applications, Music trunks in broadcast mode are recommended.

The Radius protocol that is supported on IP Trunk software is not provided for IP Peer Networking.

The use of G.723 codec can limit the number of DSP channels available on the 32-port Media Card to 24. For ITG-P Line cards, all 24 ports can be used. The use of codec G.729A/AB and G.723 impacts the voice quality, including music provided to the user.

H.323 and SIP do not support NAT. If address translation is required, it needs H.323-aware or SIP-aware NAT or VPN facilities. IP Phones (which use the proprietary UNIStim protocol) have a limited implementation of NAT.

While the CS 1000 systems supports MCDN, the systems do not support H.450 supplementary services, which is the industry-standard form of signaling used by H.323, which is equivalent to the feature transport aspect of MCDN.

Configure IP Peer Network

Contents

This section contains information on the following topics:

Overview

You use the following interfaces for configuring various components of IP Peer Networking:

- CS 1000 Element Manager
- Command Line Interface (CLI)
- NRS Manager
- Telephony Manager (TM)

> ***Note:*** You can use TM to launch Element Manager. Refer to *Telephony Manager 3.1: System Administration* (NN43050-601) for detailed information on TM.

This chapter provides instructions on how to implement IP Peer Networking in your IP Peer network using overlays and Element Manager. Once you implement the IP Peer network, you must configure data in the NRS. To configure data in the NRS refer to *Network Routing Service Installation and Commissioning (NN43001-564)* .

For information on how to install system components and how to perform basic configuration, refer to the following NTPs:

- *Communication Server 1000M and Meridian 1: Small System Installation and Commissioning* (NN43011-310)
- *Communication Server 1000M and Meridian 1: Large System Installation and Commissioning* (NN43021-310)
- *Communication Server 1000E: Installation and Commissioning* (NN43041-310)

For a description of system management, refer to *System Management* (NN43600). For a detailed description of Element Manager, refer to *Element Manager: System Administration* (NN43001-632).

Once you install the various components and configured the basic information, you then implement the IP Peer Networking feature. Implementing IP Peer Networking in a CS 1000 network is similar to configuring a traditional circuit-switched network that uses a "star" topology. All CS 1000 systems form the outer points of the star, with respect to address resolution (the systems form a grid with respect to media paths). These systems are configured to route network-wide calls into the IP network over a route configured with Virtual Trunks. The Virtual Trunks are configured to use the NRS. The NRS, in conjunction with the SIP/H.323 Gateway software at each site, acts as the center of the "star".

Element Manager and NRS Manager enable you to configure and maintain certain aspects of the system through a web browser.

> ***Note 1:*** Element Manager must be installed on each Signaling Server within the system.
>
> ***Note 2:*** Element Manager requires Internet Explorer 6.0 (or later).

In addition to Element Manager and NRS Manager, you can perform a number of configuration functions through the Command Line Interface (CLI). You can access the CLI from a serial port, or by using the Telnet or rlogin commands over a network connection.

You can also use TM to access the web server running on the Signaling Server.

Task summary

You must configure the following data when setting up an IP network:

1. Plan your Network Numbering Plan. Refer to *Dialing Plans: Description* (NN43001-283).
   a. Are you using Uniform Dialing Plan (UDP) or Coordinated Dialing Plan (CDP), or both?
   b. Are you also using Group Dialing Plan (GDP), North American Numbering Plan (NANP), or Flexible Numbering Plan (FNP)?

2. Perform basic installation, setup, and configuration of the various components, including the Signaling Server. Refer to:
   - *Communication Server 1000M and Meridian 1: Small System Installation and Commissioning* (NN43011-310)
   - *Communication Server 1000M and Meridian 1: Large System Installation and Commissioning* (NN43021-310)
   - *Communication Server 1000E: Installation and Commissioning* (NN43041-310)
   - *Signaling Server Installation and Commissioning* (NN43001-312).

3. Configure the Primary, Alternate, and Failsafe NRS at installation and initial setup of the Signaling Server. See *Signaling Server Installation and Commissioning* (NN43001-312).

   > ***Note:*** The NRS requires IP telephony node configuration files. These files are installed and configured during the Signaling Server software installation (basic configuration step).

4. Configure the Customer Data Block with any desired networking settings and options, including ISDN. Use Element Manager or the Command Line Interface (LD 15). See "Configuring the Customer Data Block"

(page 182) and "Feature Implementation of IP Peer Networking" (page 224).

5. Configure the D-channel using Element Manager or the Command Line Interface (LD 17). See "Configuring D-channels" (page 184) and "Feature Implementation of IP Peer Networking" (page 224).

6. Configure the zones.

7. Configure the SIP and/or H.323 Virtual Trunk routes using Element Manager or the Command Line Interface (LD 16). Configure the Route Data Blocks and associate the Virtual Trunk routes with the IP network by configuring the following parameters:

   a. route information

   b. network management information (for example, Access Restrictions)

   c. bandwidth zone

   d. protocol identifier

   e. associated Node ID

   For the Element Manager procedure, see "Configuring the Virtual routes and trunks" (page 189). For the CLI procedure, see "Feature Implementation of IP Peer Networking" (page 224).

8. Configure the Virtual Trunks using Element Manager (see "Configuring the Virtual routes and trunks" (page 189)) or the Command Line Interface (LD 14) and "Feature Implementation of IP Peer Networking" (page 224).

9. Use Element Manager or the Command Line Interface (CLI) to configure networking ("Configuring networking" (page 198)) and numbering plan features ("Configuring call routing" (page 202)) within the Call Server, such as routing calls based on digits dialed. For example, CDP configuration for the dialing plan used on the Call Server includes:

   a. ESN control block basics (LD 86): configure the dialing plan

   b. Network Control Block (LD 87): configure network access

   c. Route List Block (LD 86): create an entry for Virtual Trunk route

   d. Network Control Block (LD 15): enter CDP steering codes or UDP steering codes

10. Configure the codecs using Element Manager (see "Configuring codecs" (page 206)).

11. Configure dialing plan information for calls that must be routed to circuit-switched trunks (for example, PSTN interfaces). See *Dialing Plans: Description* (NN43001-283) and *IP Trunk: Description, Installation, and Operation* (NN43001-563).

12. Configure the gateways. See "Configuring the Gateways" (page 239).
    - See "Enabling and configuring the H.323 Gateway" (page 240)
    - "Enabling and configuring the SIP Trunk Gateway" (page 243)

13. Configure the NRS.

Launching Element Manager

Element Manager (EM) for CS 1000 Release 5.0 software is offered in two versions: a Linux-based EM and a VxWorks-based EM.

The Linux-based EM is a component of the Nortel Enterprise Common Manager (ECM). The ECM provides security and navigation infrastructure services for the web-based management applications: Element Manager and NRS Manager. Refer to *Enterprise Common Manager Fundamentals (NN43001-116)* for detailed information on ECM.

Element Manager is supported only on Microsoft® Internet Explorer 6.0 (or later) for the Windows® operating systems.

Linux-based Element Manager

Log in to ECM and Access Element Manager

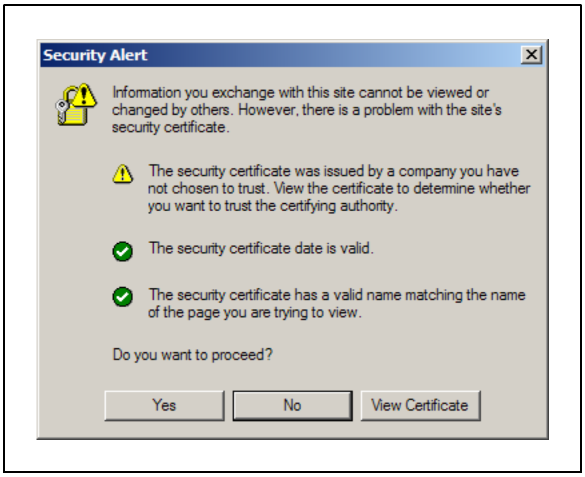
Access the Linux-based EM through the ECM. To log in to Linux-based EM follow the steps in Procedure 3 "Logging in to ECM and Accessing Linux-based Element Manager" (page 175).

Procedure 3

Logging in to ECM and Accessing Linux-based Element Manager

Step	Action
1	Enter the Fully Qualified Domain Name (FQDN), in the browser’s address field, of an ECM server. ***Note:*** The FQDN of the ECM server can be bookmarked in the Internet Explorer Favorites list. The Security Alert web page opens. See Figure 68 "Security Alert web page" (page 176).

Figure 68
Security Alert web page



Security Alert

Information you exchange with this site cannot be viewed or changed by others. However, there is a problem with the site's security certificate.

The security certificate was issued by a company you have not chosen to trust. View the certificate to determine whether you want to trust the certifying authority.

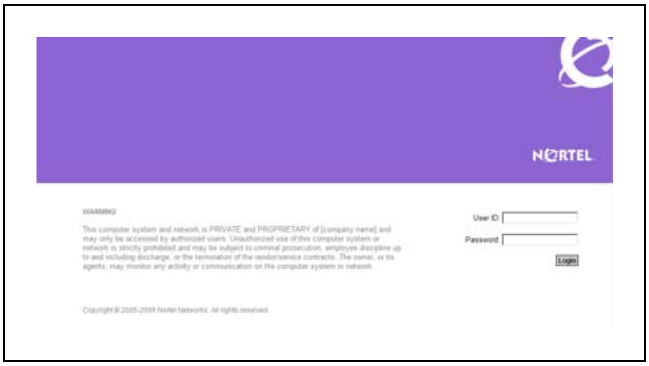
The security certificate date is valid.

The security certificate has a valid name matching the name of the page you are trying to view.

Do you want to proceed?

2 Click the **Yes** button. The ECM log in web page opens. See Figure 69 "ECM log in web page" (page 176).

Figure 69
ECM log in web page



WARNING!

This computer system and network is PRIVATE and PROPRIETARY of [company name] and may only be accessed by authorized users. Unauthorized use of this computer system or network is strictly prohibited and may be subject to criminal prosecution, employee discipline up to and including discharge, or the termination of the vendor/service contracts. The owner, or its agents, may monitor any activity or communication on the computer system or network.

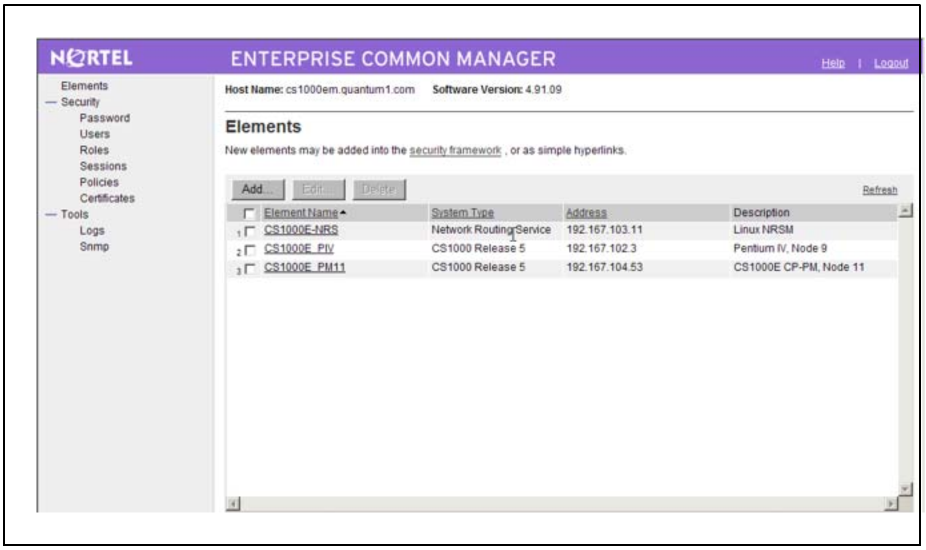
User ID
Password

Copyright © 2005-2006 Nortel Networks. All rights reserved.

3 Enter **User Name** and **Password** in the text boxes. Click the **Log in** button.

The ECM Elements web page opens. See Figure 70 "ECM Elements web page" (page 177).

Figure 70
ECM Elements web page



Host Name: cs1000em.quantum1.com **Software Version:** 4.91.09

Elements

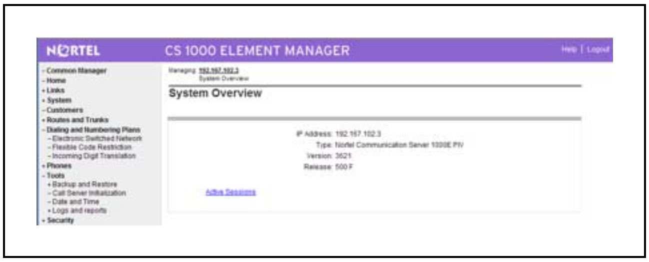
New elements may be added into the security framework , or as simple hyperlinks.

	Element Name	System Type	Address	Description
1 □	CS1000E-NRS	Network Routing Service	192.167.103.11	Linux NRSM
2 □	CS1000E_PIV	CS1000 Release 5	192.167.102.3	Pentium IV, Node 9
3 □	CS1000E_PM11	CS1000 Release 5	192.167.104.53	CS1000E CP-PM, Node 11

4 Click the link to the Element Manager in the **Element Name** column.

The Element Manager System Overview web page opens, as shown in Figure 71 "Element Manager System Overview web page" (page 177).

Figure 71
Element Manager System Overview web page



Managing: 192.167.102.3
System Overview

System Overview

IP Address: 192.167.102.3
Type: Nortel Communication Server 1000E PIV
Version: 3621
Release: 500 F

—End—

Element Manager interface

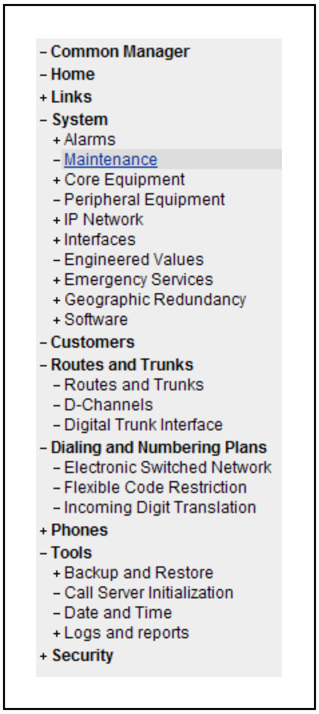
The **EM Navigator**, located on the left side of the Element Manager web pages, contains links to other web pages. **Common Manager**, the root of the EM Navigator, is a link to the ECM web page. The IP Peer Network is configured from web pages accessed through the following branches in the EM Navigator:

- Systems
- Customers

- Routes and Trunks
- Dialing and Numbering Plans
- Tools

In Figure 72 "EM Navigator" (page 178), the EM Navigator is displayed with those branches expanded.

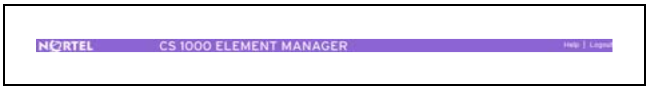
Figure 72
EM Navigator



```
- Common Manager
- Home
+ Links
- System
    + Alarms
    - Maintenance
    + Core Equipment
    - Peripheral Equipment
    + IP Network
    + Interfaces
    - Engineered Values
    + Emergency Services
    + Geographic Redundancy
    + Software
- Customers
- Routes and Trunks
    - Routes and Trunks
    - D-Channels
    - Digital Trunk Interface
- Dialing and Numbering Plans
    - Electronic Switched Network
    - Flexible Code Restriction
    - Incoming Digit Translation
+ Phones
- Tools
    + Backup and Restore
    - Call Server Initialization
    - Date and Time
    + Logs and reports
+ Security
```

Help and logout Links The **Help** and **Logout** links are located on the right side of the EM web page header. See Figure 73 "Help and Logout Links" (page 178).

Figure 73
Help and Logout Links



Help Link Select the **Help** link to access the **EM Help files** .

EM provides context-sensitive help. That is, the help page displayed depends on the EM web page from which it is opened. Once a help page is opened, click the **Show** link in the upper left corner of the page to display the **Contents** and an **Index** of the **EM Help Files**.

Logout Link Select the **Logout** link to terminate the current Enterprise Common Manager session. See Procedure 4 "Logging out of ECM" (page 179).

Common Manager link Select the **Common Manager** link on the **EM Navigator** to return to the ECM web page without terminating the current ECM session. See "Element Manager interface" (page 177).

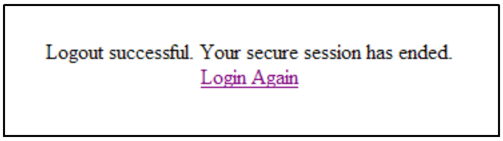
Log out of ECM

See Procedure 4 "Logging out of ECM" (page 179) to log out of the ECM. Logging out of the ECM terminates the current session.

Procedure 4
Logging out of ECM

Step	Action
1	Click the **Logout** link on the right side of the EM web page header. The **Enterprise Common Manager Logout successful** web page opens.

Figure 74
Enterprise Common Manager Logout successful web page



Logout successful. Your secure session has ended.
Login Again

Step	Action
2	Close the browser window.

—End—

VxWorks-based Element Manager

To log in to VxWorks-based EM, follow the steps in Procedure 5 "Launching VxWorks-based Element Manager" (page 179).

Procedure 5
Launching VxWorks-based Element Manager

Step	Action
1	Open the web browser.
2	Enter the **Signaling Server Node IP address** in the Address Bar of the browser window and press **Enter** on the keyboard.

Note: The ELAN network interface IP address may be required, instead of the Node IP address, to access to the Element Manager login web page in secure environments.

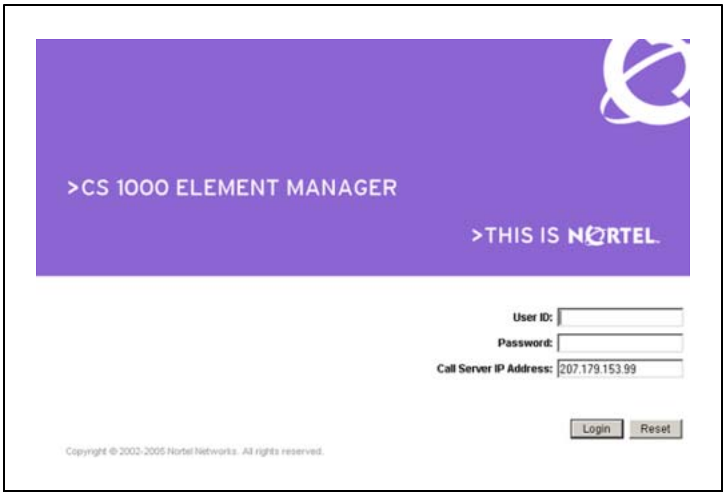
3 Element Manager launches and the **Login** web page opens (see Figure 75 "Element Manager – Login web page" (page 180)).

a. Enter the **User ID** and **Password** of the Call Server.

The IP address of the Call Server is auto-filled in the **CS IP Address** field.

b. Click **Login**.

Figure 75
Element Manager Login web page



User ID:

Password:

Call Server IP Address: 207.179.153.99

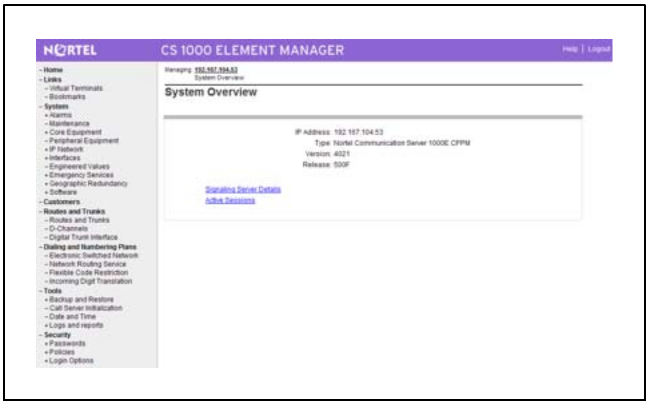
Copyright © 2002-2005 Nortel Networks. All rights reserved.

4 The **System Overview** web page opens (see Figure 76 "Element Manager – System Overview" (page 181)).

The Navigator is located on the left side of the browser window.

The **System Overview** web page contains information about the system. The web page shows that the Call Server is a central component of the system and also lists other components in the system.

Figure 76
Element Manager System Overview



Managing: 192.167.104.53
System Overview

System Overview

IP Address: 192.167.104.53
Type: Nortel Communication Server 1000E CPPM
Version: 4021
Release: 500F

Note 1: To log out of Element Manager, click **Logout** at the right in the Element Manager banner at the top of any Element Manager web page (for example, see Figure 76 "Element Manager – System Overview" (page 181)). The **Login** web page (see Figure 75 "Element Manager – Login web page" (page 180)) is displayed again. If you need to log back in to Element Manager, repeat step 3.

Note 2: Element Manager times out after a period of inactivity.

Users are logged out without any warning in all Element Manager web pages, with the exception of the **Edit** web page (see Figure 111 "Edit web page" (page 207)). When you are working in the Edit web page, a message opens that warns of the impending time-out action. Click **OK** (on the warning message) within the remaining time-out period (5 minutes) to reset the timer. If you do not respond within the 5 minute warning period, your session is canceled and you must log in again. Any data modifications made on screen, but not submitted to the system, are lost.

Note 3: For additional information about Element Manager, refer to the following NTPs:

- *Signaling Server Installation and Commissioning* (NN43001-312)
- *Element Manager: System Administration* (NN43001-632)

—End—

Using Element Manager for configuration

Read the following sections and follow the procedures in the order given.

Configuring the Customer Data Block

To configure the Customer Data Block EM, follow the steps in Procedure 6 "Configuring the Customer Data Block and enabling ISDN " (page 182).

Procedure 6
Configuring the Customer Data Block and enabling ISDN

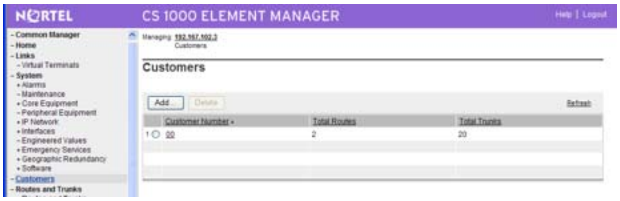
Step Action

To configure the Customer Data Block with network settings and options, you can use Element Manager or LD 15 of the Command Line Interface.

1 Click **Customers** in the EM Navigator.

The **Customers** web page opens, as shown in Figure 77 "Customers web page" (page 182)

Figure 77
Customers web page



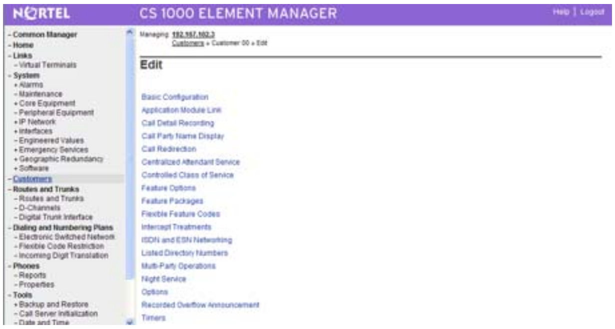
Managing: 192.167.102.3
Customers

Customers

	Customer Number ▲	Total Routes	Total Trunks
1 ○	00	2	20

2 Click a link in the **Customer Number** column. The **Edit** web page for Customer number xx opens, as shown in Figure 78 "Customer xx Edit web page" (page 182).

Figure 78
Customer xx Edit web page



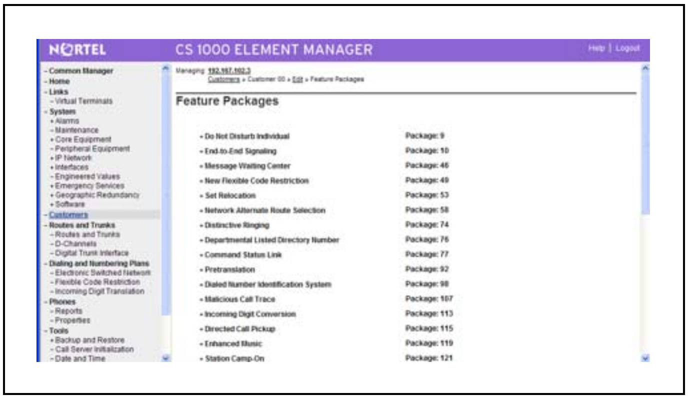
Managing: 192.167.102.3
Customers » Customer 00 » Edit

Edit

Basic Configuration
Application Module Link
Call Detail Recording
Call Party Name Display
Call Redirection
Centralized Attendant Service
Controlled Class of Service
Feature Options
Feature Packages
Flexible Feature Codes
Intercept Treatments
ISDN and ESN Networking
Listed Directory Numbers
Multi-Party Operations
Night Service
Options
Recorded Overflow Announcement
Timers

3 Click **Feature Packages**. The **Feature packages** web page opens, as shown in Figure 79 "Feature Packages web page" (page 183).

Figure 79
Feature Packages web page



NORTEL CS 1000 ELEMENT MANAGER Help | Logout

- Common Manager
- Home
- Links
  - Virtual Terminals
- System
  + Alarms
  - Maintenance
  + Core Equipment
  - Peripheral Equipment
  + IP Network
  + Interfaces
  - Engineered Values
  + Emergency Services
  + Geographic Redundancy
  + Software
- Customers
- Routes and Trunks
  - Routes and Trunks
  - D-Channels
  - Digital Trunk Interface
- Dialing and Numbering Plans
  - Electronic Switched Network
  - Flexible Code Restriction
  - Incoming Digit Translation
- Phones
  - Reports
  - Properties
- Tools
  + Backup and Restore
  - Call Server Initialization
  - Date and Time

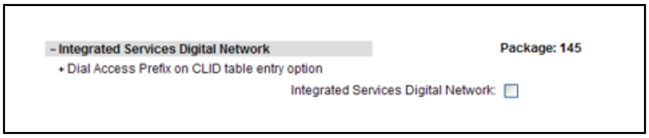
Managing: 192.167.102.3
Customers » Customer 00 » Edit » Feature Packages

Feature Packages

+ Do Not Disturb Individual	Package: 9
+ End-to-End Signaling	Package: 10
+ Message Waiting Center	Package: 46
+ New Flexible Code Restriction	Package: 49
+ Set Relocation	Package: 53
+ Network Alternate Route Selection	Package: 58
+ Distinctive Ringing	Package: 74
+ Departmental Listed Directory Number	Package: 76
+ Command Status Link	Package: 77
+ Pretranslation	Package: 92
+ Dialed Number Identification System	Package: 98
+ Malicious Call Trace	Package: 107
+ Incoming Digit Conversion	Package: 113
+ Directed Call Pickup	Package: 115
+ Enhanced Music	Package: 119
+ Station Camp-On	Package: 121

4 Scroll down the page and select **Integrated Services Digital Network Package: 145**. The **Feature packages** web page expands to display the **Integrated Services Digital Network:** check box.

Figure 80
Integrated Services Digital Network Package: 145 Expanded



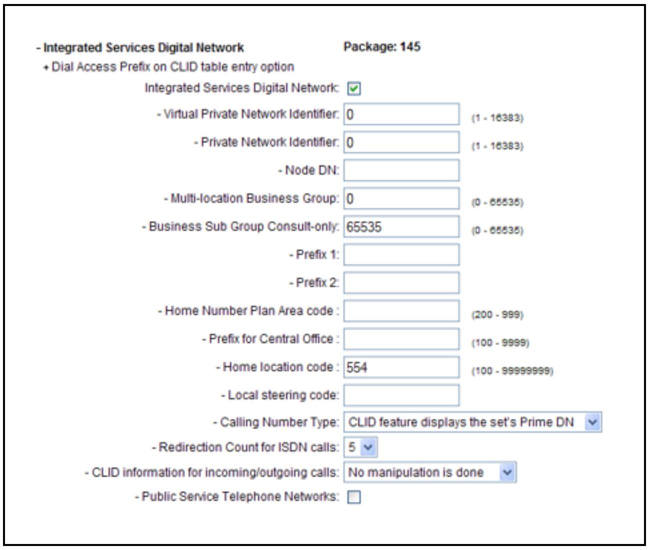
- Integrated Services Digital Network Package: 145
  + Dial Access Prefix on CLID table entry option

Integrated Services Digital Network: □

5 Click the **Integrated Services Digital Network:** check box.

The ISDN list expands to show the ISDN package options, as shown in Figure 81 "ISDN package options" (page 183).

Figure 81
ISDN package options



- Integrated Services Digital Network Package: 145
  + Dial Access Prefix on CLID table entry option

Integrated Services Digital Network: ☑
- Virtual Private Network Identifier: 0 (1 - 16383)
- Private Network Identifier: 0 (1 - 16383)
- Node DN:
- Multi-location Business Group: 0 (0 - 65535)
- Business Sub Group Consult-only: 65535 (0 - 65535)
- Prefix 1:
- Prefix 2:
- Home Number Plan Area code : (200 - 999)
- Prefix for Central Office : (100 - 9999)
- Home location code : 554 (100 - 99999999)
- Local steering code:
- Calling Number Type: CLID feature displays the set's Prime DN
- Redirection Count for ISDN calls: 5
- CLID information for incoming/outgoing calls: No manipulation is done
- Public Service Telephone Networks: □

6 Scroll to the bottom of the page and click **Save**.

—End—

Configuring D-channels

Procedure 7
Configuring D-channels

Step Action

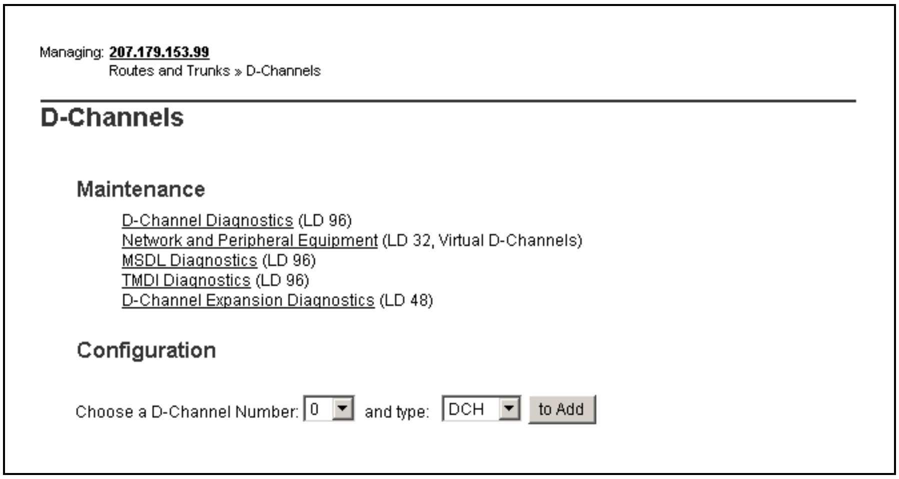
To configure D-channels, use Element Manager or LD 17 of the Command Line Interface.
*Figure 82 "D-Channels web page" (page 184) and Figure 83 "D-channels xx Property Configuration web page" (page 185) show the **D-Channel Configuration** web pages in Element Manager. Use these web pages to configure D-channels.*

1 Select **Routes and Trunks > D-Channels** from the EM Navigator.

> ***Note:*** The first time you access this web page, a message indicates that no D-channels have been configured.

The **D-Channels** web page opens as shown in Figure 82 "D-Channels web page" (page 184). This window also contains links to D-Channel maintenance and diagnostic pages.

Figure 82
D-Channels web page



Managing: 207.179.153.99
Routes and Trunks » D-Channels

D-Channels

Maintenance

D-Channel Diagnostics (LD 96)
Network and Peripheral Equipment (LD 32, Virtual D-Channels)
MSDL Diagnostics (LD 96)
TMDI Diagnostics (LD 96)
D-Channel Expansion Diagnostics (LD 48)

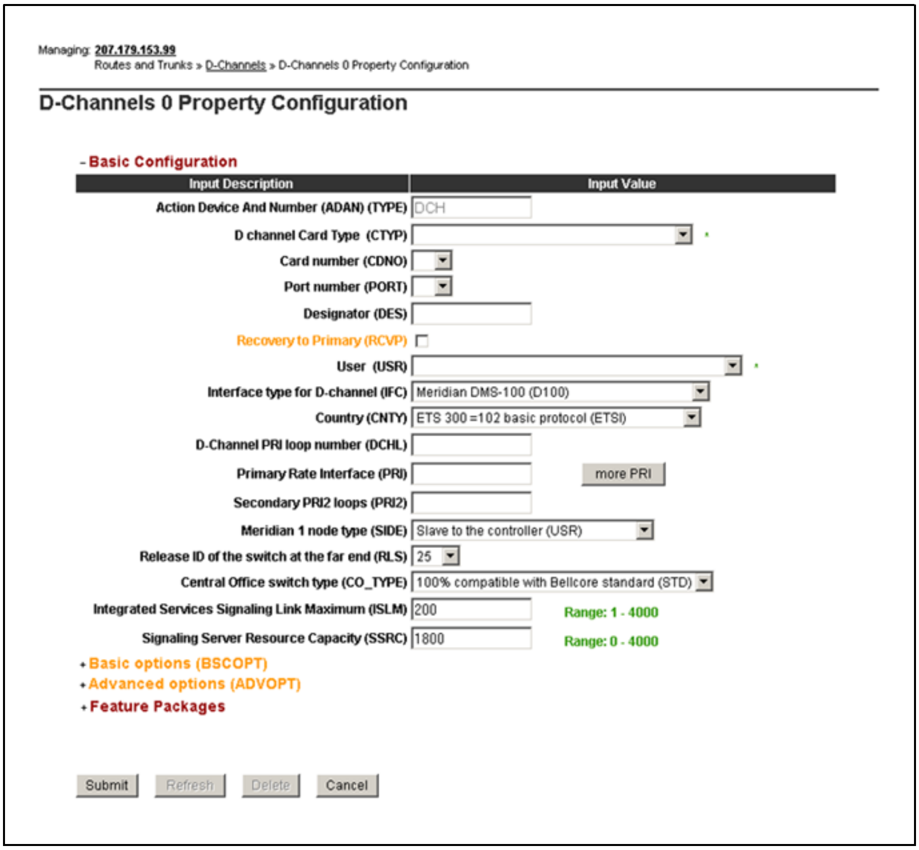
Configuration

Choose a D-Channel Number: 0 and type: DCH

2 In the **Configuration** section, input the D-channel number and type. Click **to Add**.

The **D-Channels xx Property Configuration** web page opens, as shown in Figure 83 "D-channels xx Property Configuration web page" (page 185). The D-channel number is denoted by xx. Required fields are indicated with a green asterisk.

Figure 83
D-channels xx Property Configuration web page



Managing: 207.179.153.99
Routes and Trunks » D-Channels » D-Channels 0 Property Configuration

D-Channels 0 Property Configuration

- Basic Configuration

Input Description	Input Value	
Action Device And Number (ADAN) (TYPE)	DCH	
D channel Card Type (CTYP)		*
Card number (CDNO)		
Port number (PORT)		
Designator (DES)		
Recovery to Primary (RCVP)	□	
User (USR)		*
Interface type for D-channel (IFC)	Meridian DMS-100 (D100)	
Country (CNTY)	ETS 300 =102 basic protocol (ETSI)	
D-Channel PRI loop number (DCHL)		
Primary Rate Interface (PRI)		more PRI
Secondary PRI2 loops (PRI2)		
Meridian 1 node type (SIDE)	Slave to the controller (USR)	
Release ID of the switch at the far end (RLS)	25	
Central Office switch type (CO_TYPE)	100% compatible with Bellcore standard (STD)	
Integrated Services Signaling Link Maximum (ISLM)	200	Range: 1 - 4000
Signaling Server Resource Capacity (SSRC)	1800	Range: 0 - 4000

3 Configure the following fields with the following values:

a. **D channel Card Type (CYTP)** = D-Channel is over IP (DCIP)

b. **User (USR)** = Integrated Services Signaling Link Dedicated (ISLD)

c. **Interface type for D-channel (IFC)** = Meridian Meridian1 (SL1)

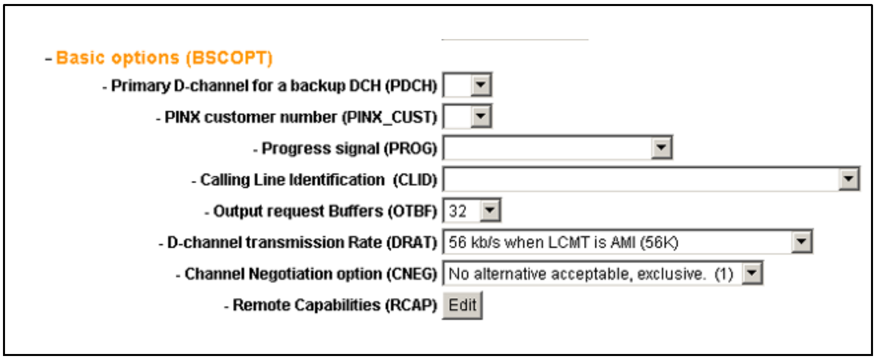
4 If you are defining the Network Name Display:

a. Select the **Release ID of the switch at the far end (RLS)** from the drop-down list.

b. Click **Basic options (BSCOPT)** tab.

The **Basic options** list expands, as shown in Figure 84 "D-channel — Basic options" (page 186).

Figure 84
D-channel Basic options



- Basic options (BSCOPT)
- Primary D-channel for a backup DCH (PDCH)
- PINX customer number (PINX_CUST)
- Progress signal (PROG)
- Calling Line Identification (CLID)
- Output request Buffers (OTBF) 32
- D-channel transmission Rate (DRAT) 56 kb/s when LCMT is AMI (56K)
- Channel Negotiation option (CNEG) No alternative acceptable, exclusive. (1)
- Remote Capabilities (RCAP) Edit

c. Configure **Remote Capabilities (RCAP)** by clicking **Edit**.

The **Remote Capabilities Configuration** web page opens.

d. Scroll down the page and click the check box for **Network name Display method 2 (ND2)**.

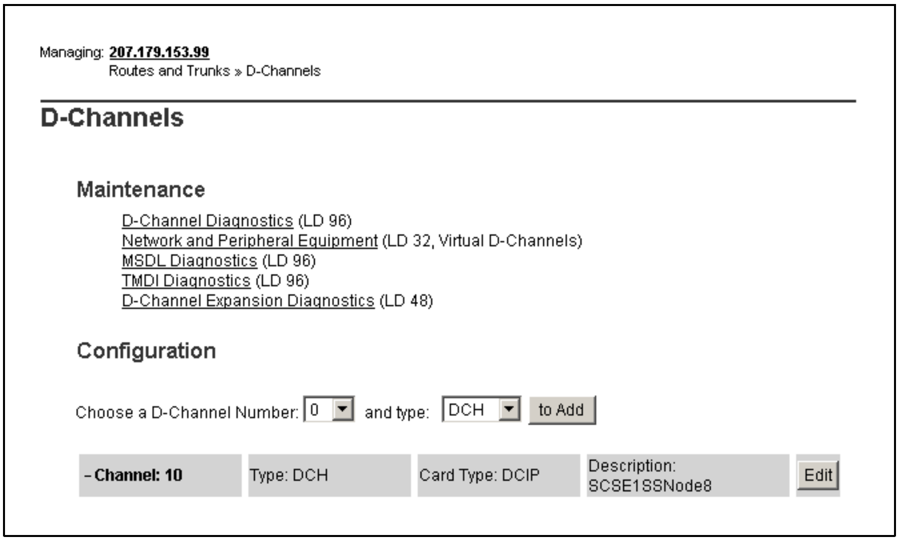
e. Click **Return - Remote Capabilities** at the bottom of the page.

The **D-Channel xx Property Configuration** web page reopens.

5 Click **Submit** to save the changes.

The **D-Channels** web page reopens (Figure 85 "D-channel configuration results" (page 186)) with the changes.

Figure 85
D-channel configuration results



Managing: 207.179.153.99
Routes and Trunks » D-Channels

D-Channels

Maintenance

D-Channel Diagnostics (LD 96)
Network and Peripheral Equipment (LD 32, Virtual D-Channels)
MSDL Diagnostics (LD 96)
TMDI Diagnostics (LD 96)
D-Channel Expansion Diagnostics (LD 48)

Configuration

Choose a D-Channel Number: 0 and type: DCH to Add

- Channel: 10	Type: DCH	Card Type: DCIP	Description: SCSE1SSNode8	Edit

—End—

Configuring zones

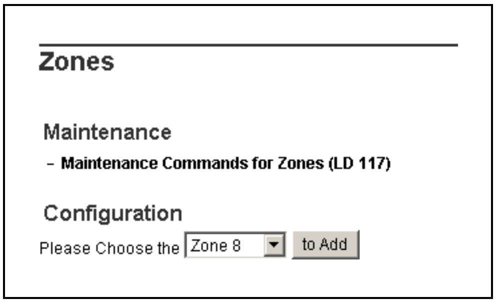
A zone is an area of a network that can be treated as a single entity with respect to the use of bandwidth for voice and signaling. Zones must be configured before the configuration of virtual routes.

Procedure 8
Configuring zones

Step	Action
1	Select **System > IP Network > Zones** from the EM Navigator.

The **Zones** web page opens (see Figure 86 "Zones web page" (page 187)). This page also contains a link to Maintenance Commands for Zones, using LD 117.

Figure 86
Zones web page



Zones

Maintenance

– Maintenance Commands for Zones (LD 117)

Configuration

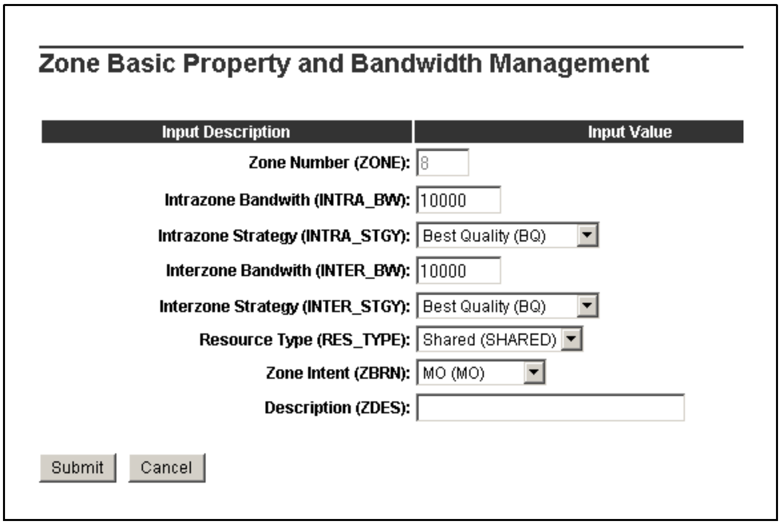
Please Choose the Zone 8 ▼ to Add

2 Choose a zone number from the drop-down list.

3 Click **to Add**.

4 The **Zone Basic Property and Bandwidth Management** web page opens (see Figure 87 "Zone Basic Property and Bandwidth Management web page" (page 188)).

Figure 87
Zone Basic Property and Bandwidth Management web page



Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	8
Intrazone Bandwith (INTRA_BW):	10000
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwith (INTER_BW):	10000
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	MO (MO)
Description (ZDES):	

Note: The **Zone Number (ZONE)** field is auto-filled based on the number selected on the Zone List web page.

5 Enter the **Intrazone Bandwidth (INTRA _BW)**.

6 Select the **Intrazone Strategy (INTRA _STGY)** from the drop-down list.

7 Enter the **Interzone Bandwidth (INTER _BW)**.

8 Select the **Interzone Bandwidth (INTER _BW)** from the drop-down list.

9 Select the **Resource Type (RES_TYPE)** from the drop-down list.

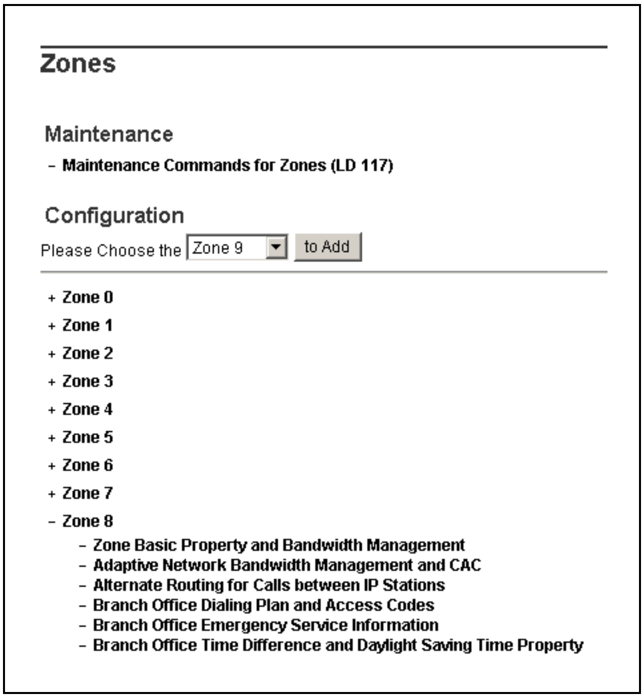
10 Select the **Branch Office Support (ZBRN)** from the drop-down list.

11 Enter a description of the zone in the **Description (ZDES)** text box.

12 Click **Submit**.

The **Zones** web page reopens with the new zone added (see Figure 88 "Zones web page with newly added zone" (page 189)).

Figure 88
Zones web page with newly added zone



Zones

Maintenance

- Maintenance Commands for Zones (LD 117)

Configuration

Please Choose the Zone 9 to Add

+ Zone 0
+ Zone 1
+ Zone 2
+ Zone 3
+ Zone 4
+ Zone 5
+ Zone 6
+ Zone 7
- Zone 8
  - Zone Basic Property and Bandwidth Management
  - Adaptive Network Bandwidth Management and CAC
  - Alternate Routing for Calls between IP Stations
  - Branch Office Dialing Plan and Access Codes
  - Branch Office Emergency Service Information
  - Branch Office Time Difference and Daylight Saving Time Property

—End—

Configuring the Virtual routes and trunks

To configure Virtual Trunk routes, you can use Element Manager or LD 16 of the Command Line Interface.

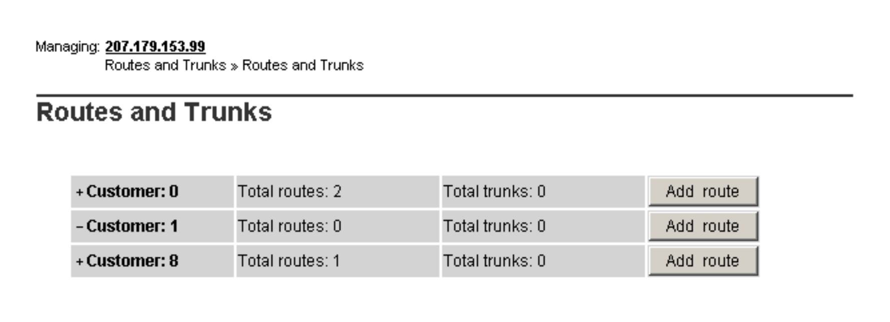
Figure 90 "New Route Configuration web page" (page 190) shows the **New Route Configuration** web page in Element Manager. Use this web page to configure Virtual Trunk routes.

> ***Note:*** The zone parameter makes the codec selections and calculates the bandwidth usage for calls to the trunk members of a given route.

Procedure 9
Configuring Virtual Trunk routes

Step	Action
1	Select **Routes and Trunks > Routes and Trunks** from the EM Navigator. The **Routes and Trunks** web page opens, as shown in Figure 89 "Routes and Trunks web page" (page 190).

Figure 89
Routes and Trunks web page



Managing: 207.179.153.99
Routes and Trunks » Routes and Trunks

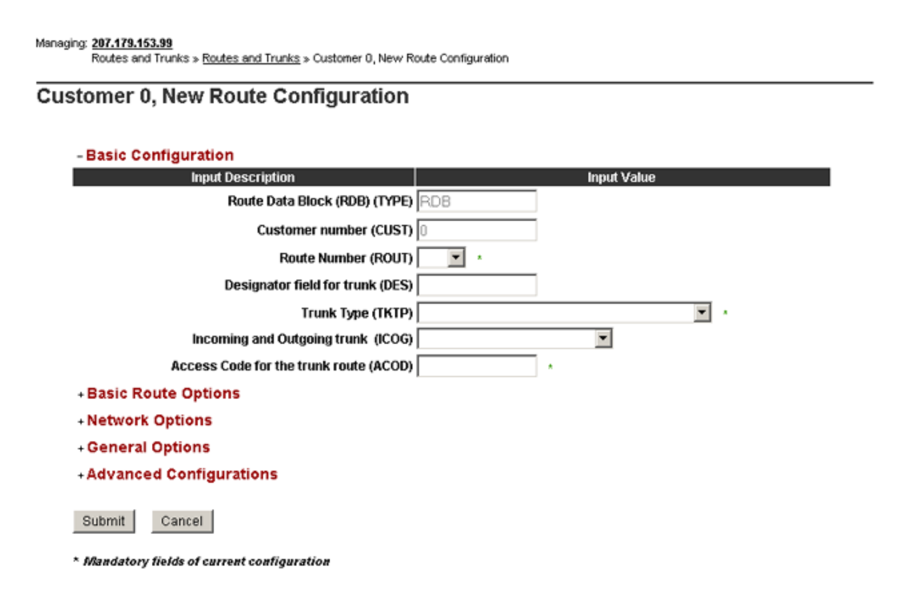
Routes and Trunks

+ Customer: 0	Total routes: 2	Total trunks: 0	Add route
- Customer: 1	Total routes: 0	Total trunks: 0	Add route
+ Customer: 8	Total routes: 1	Total trunks: 0	Add route

2 Click **Add route** associated with the customer.

The **Customer xx, New Route Configuration** web page opens (where xx is the customer number). See Figure 90 "New Route Configuration web page" (page 190).

Figure 90
New Route Configuration web page



Managing: 207.179.153.99
Routes and Trunks » Routes and Trunks » Customer 0, New Route Configuration

Customer 0, New Route Configuration

- Basic Configuration

Input Description	Input Value
Route Data Block (RDB) (TYPE)	RDB
Customer number (CUST)	0
Route Number (ROUT)	*
Designator field for trunk (DES)	
Trunk Type (TKTP)	*
Incoming and Outgoing trunk (ICOG)	
Access Code for the trunk route (ACOD)	*

+ Basic Route Options
+ Network Options
+ General Options
+ Advanced Configurations

Submit Cancel

* *Mandatory fields of current configuration*

3 Under **Basic Configuration**, fill in the required fields to create a new Virtual Trunk Route:

a. Select a **Route Number (ROUT)** from the drop-down list.

b. Select the **Trunk Type (TKTP)** = TIE trunk data block (TIE).

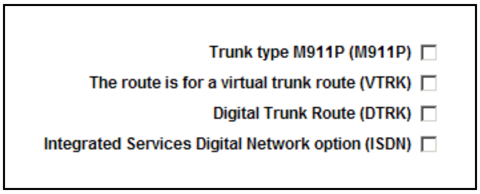
When **Trunk Type (TKTP)** is selected, the following three options appear (see Figure 91 "Options available when TIE is selected" (page 191)):

- **The route is for a virtual trunk route (VTRK)** (see step 4)

- **Digital Trunk Route (DTRK)**
- **Integrated Services Digital Network option (ISDN)** (see step 5)

c. Enter the **Access Code for the trunk route (ACOD)**.

Figure 91
Options available when TIE is selected

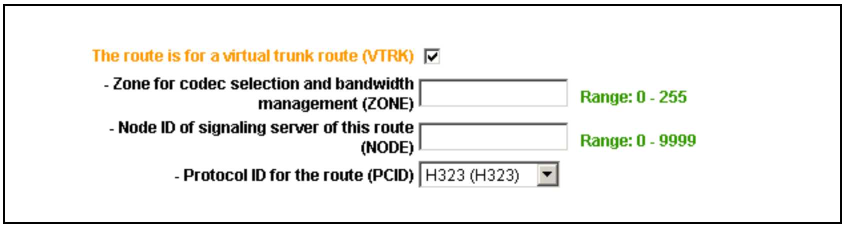


Trunk type M911P (M911P) □
The route is for a virtual trunk route (VTRK) □
Digital Trunk Route (DTRK) □
Integrated Services Digital Network option (ISDN) □

4 Select **The route is for a virtual trunk route (VTRK)** check box.

Three fields display as shown in Figure 92 "Virtual trunk route" (page 191).

Figure 92
Virtual trunk route



The route is for a virtual trunk route (VTRK) ☑
- Zone for codec selection and bandwidth management (ZONE) [ ] Range: 0 - 255
- Node ID of signaling server of this route (NODE) [ ] Range: 0 - 9999
- Protocol ID for the route (PCID) H323 (H323) ▾

a. Enter a **ZONE** number.

b. Enter the **NODE** ID (the node served by this Signaling Server).

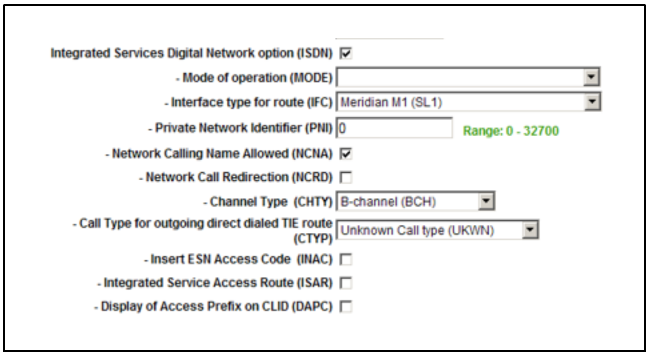
c. Select the **Protocol ID for the route (PCID)**. H323 (H323) and SIP (SIP) are two of the available options.

Note: If SIP is selected as the protocol ID for the route (PCID), then the **Print Correlation ID in CDR for the route (CRID)** check box is displayed. CRID only appears if VTRK is YES and PCID is SIP and CDR is turned on for the route.

5 Select the **Integrated Services Digital Networks option (ISDN)** check box.

The ISDN section expands as shown in Figure 93 "ISDN option" (page 192).

Figure 93
ISDN option



Integrated Services Digital Network option (ISDN)	☑
- Mode of operation (MODE)	
- Interface type for route (IFC)	Meridian M1 (SL1)
- Private Network Identifier (PNI)	0 Range: 0 - 32700
- Network Calling Name Allowed (NCNA)	☑
- Network Call Redirection (NCRD)	□
- Channel Type (CHTY)	B-channel (BCH)
- Call Type for outgoing direct dialed TIE route (CTYP)	Unknown Call type (UKWN)
- Insert ESN Access Code (INAC)	□
- Integrated Service Access Route (ISAR)	□
- Display of Access Prefix on CLID (DAPC)	□

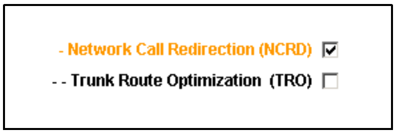
a. Choose **Mode of operations (MODE)** = Route uses ISDN Signaling Link (ISLD).

b. Choose **Interface type for route (IFC)** = Meridian M1 (SL1).

c. Select the **Network Calling Name Allowed (NCNA)** check box.

6 Select the **Network Call Redirection (NCRD)** check box (see Figure 94 "NCRD" (page 192)).

Figure 94
NCRD

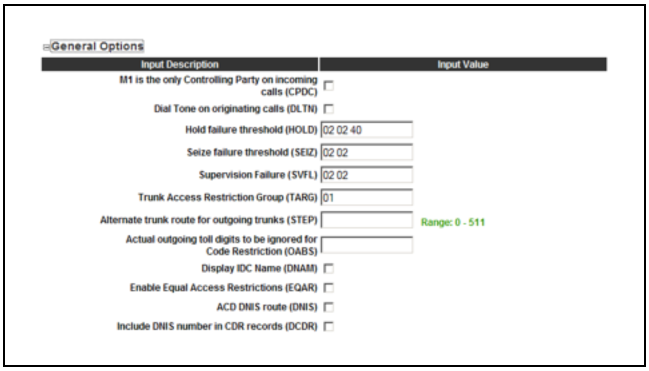


- Network Call Redirection (NCRD)	☑
- - Trunk Route Optimization (TRO)	□

7 Click **General Options**.

The General Options list expands, as shown in Figure 95 "General Options" (page 192).

Figure 95
General Options



⊟General Options

Input Description	Input Value
M1 is the only Controlling Party on incoming calls (CPDC)	□
Dial Tone on originating calls (DLTN)	□
Hold failure threshold (HOLD)	02 02 40
Seize failure threshold (SEIZ)	02 02
Supervision Failure (SVFL)	02 02
Trunk Access Restriction Group (TARG)	01
Alternate trunk route for outgoing trunks (STEP)	Range: 0 - 511
Actual outgoing toll digits to be ignored for Code Restriction (OABS)	
Display IDC Name (DNAM)	□
Enable Equal Access Restrictions (EQAR)	□
ACD DNIS route (DNIS)	□
Include DNIS number in CDR records (DCDR)	□

8 Enter the **Trunk Access Restriction Group (TARG)** value if you are configuring a single customer.

9 Enter the appropriate information in the text boxes and in **Basic Route Options, Network Options, General Options**, and **Advanced Configurations**.

10 Click **Submit**.

The **Trunks and Routes** web page opens and the newly configured route is displayed for the customer.

—End—

Configure virtual superloops for IP Phones (LD 97)

One or more virtual superloops must be configured to support IP Phone Virtual TNs (VTNs).

Large Systems

IIn Large Systems, virtual superloops contend for the same range of loops with phantom, standard and remote superloops, digital trunk loops and all service loops. Virtual superloops can reside in physically-equipped network groups or in virtual network groups.

A 61c is a single group machine and can have physical loops 0-31 and virtual loops up to 159.

An 81c is a multi-group machine and can have physical and virtual loops 0-159. An 81c with the FIBN package and FIBN hardware can have physical and virtual loop 0-255.

Virtual superloops have 1024 TNs and are non-blocking. Therefore all 1024 TNs can be configured on a virtual superloop and still be a non-blocking configuration. Virtual Superloops are configured in LD 97.

LD 97 Configure virtual superloop for Large Systems.

Prompt	Response	Description
REQ	CHG	Change

Prompt	Response	Description
TYPE	SUPL	Superloop
SUPL	Vxxx	V stands for a virtual superloop and xxx is the number of the virtual superloop xxx = 0 – 156 in multiples of four for a Large System without Fiber Network Package (FIBN) package 365 xxx = 0 – 252 in multiples of four for a Large System with Fiber Network Package (FIBN) package 365 xxx = 0 – 252 in multiples of four for a CS 1000E system xxx = 96 – 112 in multiples of four for a Small System

CS 1000E systems

Table 12 "Virtual superloop/virtual card mapping for CS 1000E Systems" (page 194) lists the virtual superloop and virtual card mapping for the CS 1000E system.

Table 12
Virtual superloop/virtual card mapping for CS 1000E Systems

SUPL	Card	
96	61-64	81-84
100	65-68	85-88
104	69-72	89-92
108	73-76	93-96
112	77-80	97-99

LD 97 PRT TYPE SUPL prints the implicit virtual, phantom, or DECT cards for a virtual, phantom, or DECT superloop.

LD 21 LUU allows the user to list unused units of a specified type (iset, vtrk, phantom, DECT) in a specified range of (virtual, and so on) TNs. Similarly, LUC of a specified type (virtual, phantom, or DECT) prints a list of unused cards on configured superloops.

Procedure 10
Configuring Virtual Trunks

Step Action

To configure Virtual Trunks in Element Manager, use the "New Member Property" pages.

Figure 97 "New Trunk Configuration web page" (page 196) to Figure 99 "New Trunk Configuration – Advanced Trunk Configurations" (page 198) show the New Member Property web page in Element Manager. Use this web page to configure Virtual Trunks.

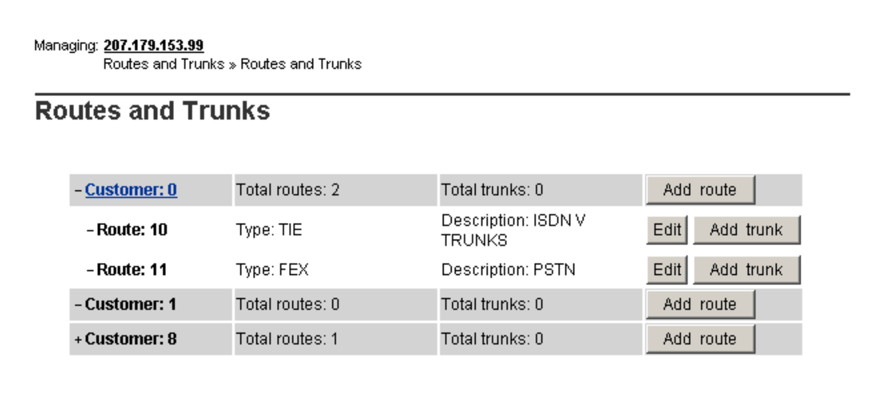
1 Select **Routes and Trunks > Routes and Trunks** from the EM Navigator.

The **Routes and Trunks** web pages opens (see Figure 89 "Routes and Trunks web page" (page 190)).

2 Select the **Customer** for which you are configuring Virtual Trunks.

The customer list expands showing a list of configured routes, as shown in Figure 96 "Customer routes" (page 195).

Figure 96
Customer routes



Managing: 207.179.153.99
Routes and Trunks » Routes and Trunks

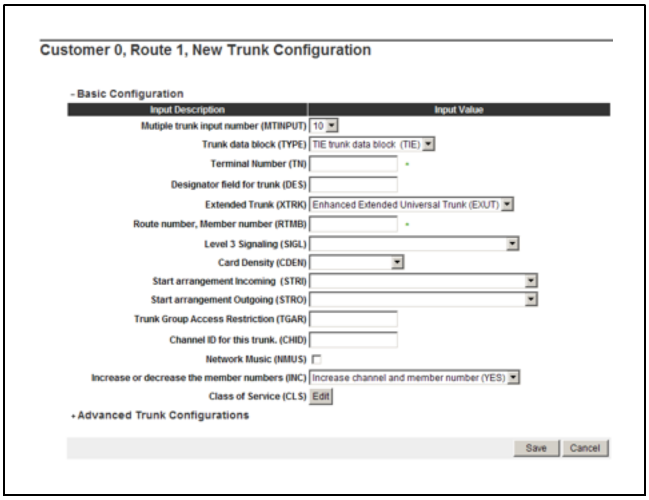
Routes and Trunks

- Customer: 0	Total routes: 2	Total trunks: 0	Add route
- Route: 10	Type: TIE	Description: ISDN V TRUNKS	Edit Add trunk
- Route: 11	Type: FEX	Description: PSTN	Edit Add trunk
- Customer: 1	Total routes: 0	Total trunks: 0	Add route
+ Customer: 8	Total routes: 1	Total trunks: 0	Add route

3 Click **Add trunk** associated with the route listing to add new trunk members.

The **Customer xx, Route yy, New Trunk Configuration** web page opens, as shown in Figure 97 "New Trunk Configuration web page" (page 196).

Figure 97
New Trunk Configuration web page



Customer 0, Route 1, New Trunk Configuration

- Basic Configuration

Input Description	Input Value
Mutiple trunk input number (MTINPUT)	10 ▾
Trunk data block (TYPE)	TIE trunk data block (TIE) ▾
Terminal Number (TN)	*
Designator field for trunk (DES)	
Extended Trunk (XTRK)	Enhanced Extended Universal Trunk (EXUT) ▾
Route number, Member number (RTMB)	*
Level 3 Signaling (SIGL)	▾
Card Density (CDEN)	▾
Start arrangement Incoming (STRI)	▾
Start arrangement Outgoing (STRO)	▾
Trunk Group Access Restriction (TGAR)	
Channel ID for this trunk. (CHID)	
Network Music (NMUS)	□
Increase or decrease the member numbers (INC)	Increase channel and member number (YES) ▾
Class of Service (CLS)	Edit

+ Advanced Trunk Configurations

4 Choose **Multiple trunk input number (MTINPUT)** if you are using more than one trunk.

5 Select **Trunk data block (TYPE)** = IP Trunk (IPTI).

6 **Terminal Number (TN)**.

7 (Optional) **Designator field for trunk (DES)** is a text string only, and has no impact on functionality.

8 Select **Extended Trunk (XTRK)** option.

9 Enter a **Route number, Member number (RTMB)**.

10 Enter a **Trunk Group Access Restriction (TGAR) value**.

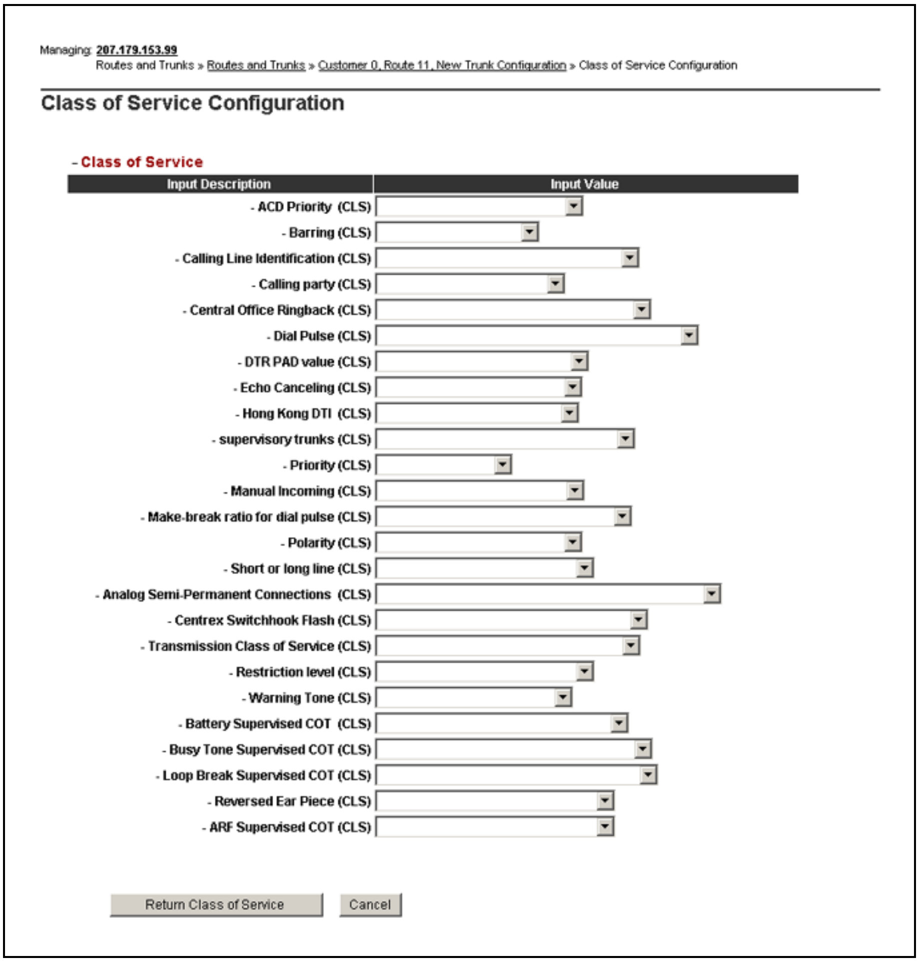
11 Enter a **Channel ID for this trunk (CHID)** = x (where x is in the range of 1-382).

> ***Note:*** Channel_ID: A numeric input is required. However, there is no requirement for the CHID of Site A to match the CHID of Site B, as required with traditional ISL trunking as the channel is no longer point-to-point.

12 To specify a **Class of Service (CLS)** for the trunk, click **Edit**.

The **Class of Service Configuration** web page opens (see Figure 98 "New Trunk Configuration – Class of Service Configuration web page" (page 197)). Select a Class of Service.

Figure 98
New Trunk Configuration Class of Service Configuration web page



Managing: 207.179.153.99
Routes and Trunks » Routes and Trunks » Customer 0, Route 11, New Trunk Configuration » Class of Service Configuration

Class of Service Configuration

- Class of Service

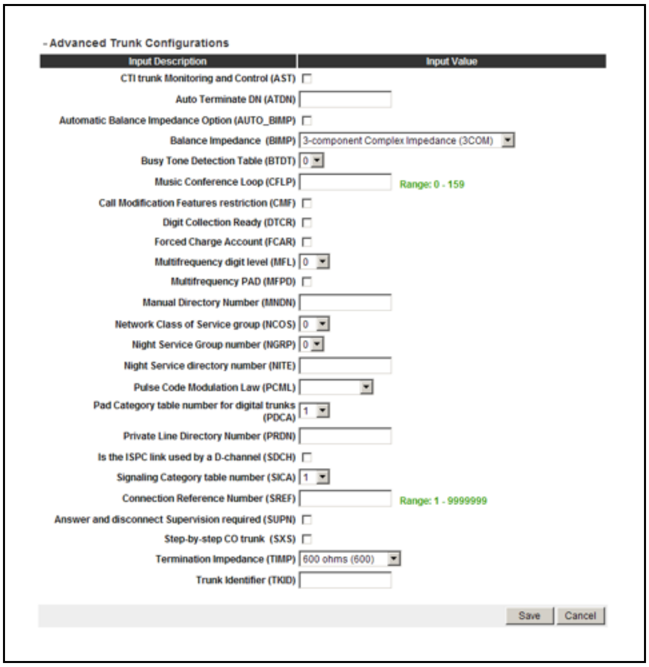
Input Description	Input Value
- ACD Priority (CLS)	
- Barring (CLS)	
- Calling Line Identification (CLS)	
- Calling party (CLS)	
- Central Office Ringback (CLS)	
- Dial Pulse (CLS)	
- DTR PAD value (CLS)	
- Echo Canceling (CLS)	
- Hong Kong DTI (CLS)	
- supervisory trunks (CLS)	
- Priority (CLS)	
- Manual Incoming (CLS)	
- Make-break ratio for dial pulse (CLS)	
- Polarity (CLS)	
- Short or long line (CLS)	
- Analog Semi-Permanent Connections (CLS)	
- Centrex Switchhook Flash (CLS)	
- Transmission Class of Service (CLS)	
- Restriction level (CLS)	
- Warning Tone (CLS)	
- Battery Supervised COT (CLS)	
- Busy Tone Supervised COT (CLS)	
- Loop Break Supervised COT (CLS)	
- Reversed Ear Piece (CLS)	
- ARF Supervised COT (CLS)	

13 Select the Class of Service and then click **Return Class of Service** to return to the **New Trunk Configuration** web page (see Figure 97 "New Trunk Configuration web page" (page 196)).

14 Select **Advanced Trunk Configurations**.

The **Advanced Trunk Configurations** list expands, as shown in Figure 99 "New Trunk Configuration – Advanced Trunk Configurations" (page 198).

Figure 99
New Trunk Configuration Advanced Trunk Configurations



- Advanced Trunk Configurations

Input Description	Input Value	
CTI trunk Monitoring and Control (AST)	□	
Auto Terminate DN (ATDN)		
Automatic Balance Impedance Option (AUTO_BIMP)	□	
Balance Impedance (BIMP)	3-component Complex Impedance (3COM)	
Busy Tone Detection Table (BTDT)	0	
Music Conference Loop (CFLP)		Range: 0 - 159
Call Modification Features restriction (CMF)	□	
Digit Collection Ready (DTCR)	□	
Forced Charge Account (FCAR)	□	
Multifrequency digit level (MFL)	0	
Multifrequency PAD (MFPD)	□	
Manual Directory Number (MNDN)		
Network Class of Service group (NCOS)	0	
Night Service Group number (NGRP)	0	
Night Service directory number (NITE)		
Pulse Code Modulation Law (PCML)		
Pad Category table number for digital trunks (PDCA)	1	
Private Line Directory Number (PRDN)		
Is the ISPC link used by a D-channel (SDCH)	□	
Signaling Category table number (SICA)	1	
Connection Reference Number (SREF)		Range: 1 - 9999999
Answer and disconnect Supervision required (SUPN)	□	
Step-by-step CO trunk (SXS)	□	
Termination Impedance (TIMP)	600 ohms (600)	
Trunk Identifier (TKID)		

15 Configure **Network Class of Service group (NCOS)**.

16 Click **Save** to save the changes.

The **Customer Explorer** web page reopens, showing the new trunk member.

—End—

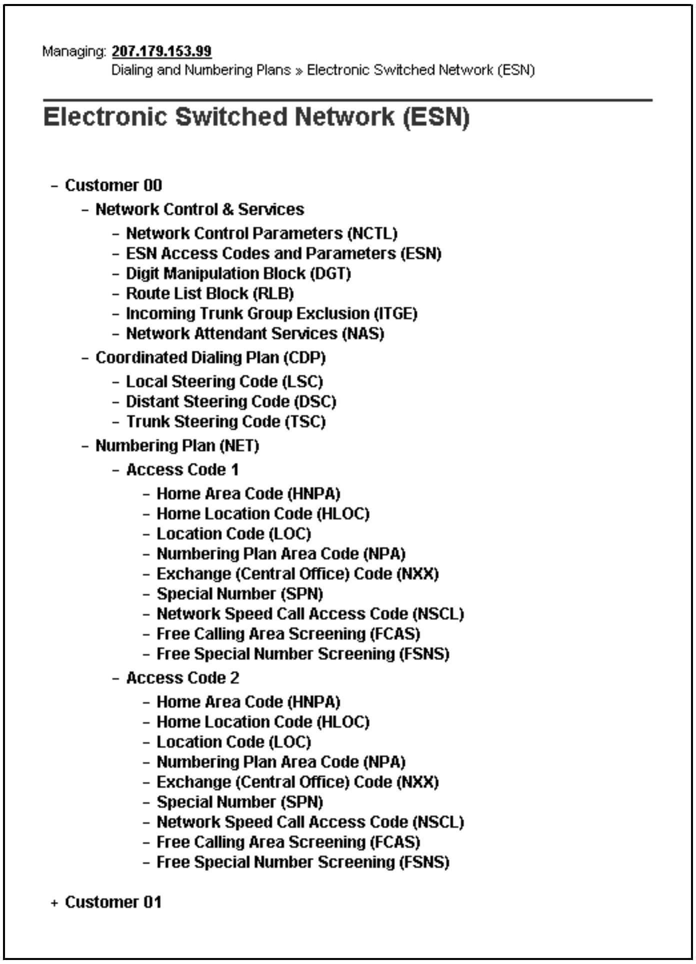
Configuring networking

The following procedures indicate a Coordinated Dialing Plan for the configuration of networking.

Procedure 11
Creating an ESN control block

Step	Action
1	Select **Dialing and Numbering Plans > Electronic Switched Network** from the EM Navigator. The **Electronic Switched Network (ESN)** web page opens, as shown in Figure 100 "Electronic Switched Network (ESN) web page" (page 199).

Figure 100
Electronic Switched Network (ESN) web page



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN)

Electronic Switched Network (ESN)

- Customer 00
  - Network Control & Services
    - Network Control Parameters (NCTL)
    - ESN Access Codes and Parameters (ESN)
    - Digit Manipulation Block (DGT)
    - Route List Block (RLB)
    - Incoming Trunk Group Exclusion (ITGE)
    - Network Attendant Services (NAS)
  - Coordinated Dialing Plan (CDP)
    - Local Steering Code (LSC)
    - Distant Steering Code (DSC)
    - Trunk Steering Code (TSC)
  - Numbering Plan (NET)
    - Access Code 1
      - Home Area Code (HNPA)
      - Home Location Code (HLOC)
      - Location Code (LOC)
      - Numbering Plan Area Code (NPA)
      - Exchange (Central Office) Code (NXX)
      - Special Number (SPN)
      - Network Speed Call Access Code (NSCL)
      - Free Calling Area Screening (FCAS)
      - Free Special Number Screening (FSNS)
    - Access Code 2
      - Home Area Code (HNPA)
      - Home Location Code (HLOC)
      - Location Code (LOC)
      - Numbering Plan Area Code (NPA)
      - Exchange (Central Office) Code (NXX)
      - Special Number (SPN)
      - Network Speed Call Access Code (NSCL)
      - Free Calling Area Screening (FCAS)
      - Free Special Number Screening (FSNS)

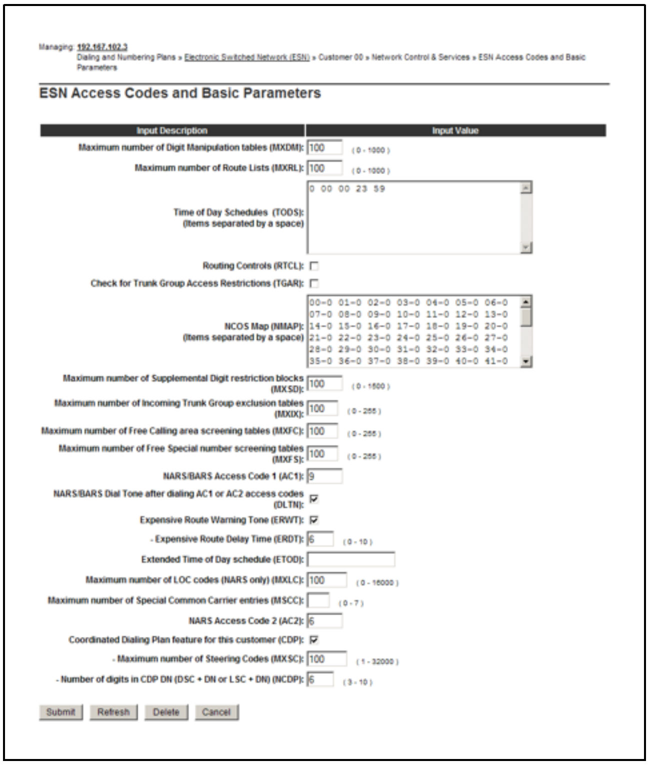
+ Customer 01

2 Under Network Control & Service, click **ESN Access Codes and Parameters (ESN)**.

If no ESN database is configured, a warning dialog box opens. Click **OK** on the warning dialog box.

The **ESN Access Codes and Basic Parameters** web page opens, as shown in Figure 101 "ESN Access Codes and Basic Parameters web page" (page 200).

Figure 101
ESN Access Codes and Basic Parameters web page



Managing: 192.167.102.3
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » ESN Access Codes and Basic Parameters

ESN Access Codes and Basic Parameters

Input Description	Input Value
Maximum number of Digit Manipulation tables (MXDM):	100 (0 - 1000)
Maximum number of Route Lists (MXRL):	100 (0 - 1000)
Time of Day Schedules (TODS): (Items separated by a space)	0 00 00 23 59
Routing Controls (RTCL):	☐
Check for Trunk Group Access Restrictions (TGAR):	☐
NCOS Map (NMAP): (Items separated by a space)	00-0 01-0 02-0 03-0 04-0 05-0 06-0 07-0 08-0 09-0 10-0 11-0 12-0 13-0 14-0 15-0 16-0 17-0 18-0 19-0 20-0 21-0 22-0 23-0 24-0 25-0 26-0 27-0 28-0 29-0 30-0 31-0 32-0 33-0 34-0 35-0 36-0 37-0 38-0 39-0 40-0 41-0
Maximum number of Supplemental Digit restriction blocks (MXSD):	100 (0 - 1500)
Maximum number of Incoming Trunk Group exclusion tables (MXIX):	100 (0 - 255)
Maximum number of Free Calling area screening tables (MXFC):	100 (0 - 255)
Maximum number of Free Special number screening tables (MXFS):	100 (0 - 255)
NARS/BARS Access Code 1 (AC1):	9
NARS/BARS Dial Tone after dialing AC1 or AC2 access codes (DLTN):	☑
Expensive Route Warning Tone (ERWT):	☑
- Expensive Route Delay Time (ERDT):	6 (0 - 10)
Extended Time of Day schedule (ETOD):	
Maximum number of LOC codes (NARS only) (MXLC):	100 (0 - 16000)
Maximum number of Special Common Carrier entries (MSCC):	(0 - 7)
NARS Access Code 2 (AC2):	6
Coordinated Dialing Plan feature for this customer (CDP):	☑
- Maximum number of Steering Codes (MXSC):	100 (1 - 32000)
- Number of digits in CDP DN (DSC + DN or LSC + DN) (NCDP):	6 (3 - 10)

3 Define the parameters for the network. Include the **Maximum number of Route Lists (MXRL)**.

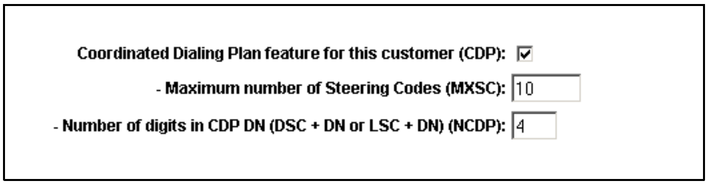
4 Scroll down the page and select the **Coordinated Dialing Plan feature for this customer (CDP)** check box.

The CDP list expands, as shown in Figure 102 "ESN data block configuration – Coordinated Dialing Plan" (page 200).

a. Configure the number of CDP steering codes (**Maximum number of Steering Codes (MXSC)**).

b. Configure the number of digits of the CDP dialed number (**Number of digits in CDP DN (DSC + DN or LSC + DN) (NCDP)**).

Figure 102
ESN data block configuration Coordinated Dialing Plan



Coordinated Dialing Plan feature for this customer (CDP):	☑
- Maximum number of Steering Codes (MXSC):	10
- Number of digits in CDP DN (DSC + DN or LSC + DN) (NCDP):	4

5 Click **Submit** to save the changes.

The **Electronic Switched Network (ESN)** web page reopens (Figure 100 "Electronic Switched Network (ESN) web page" (page 199)).

—End—

Procedure 12
Configuring network access

Step	Action

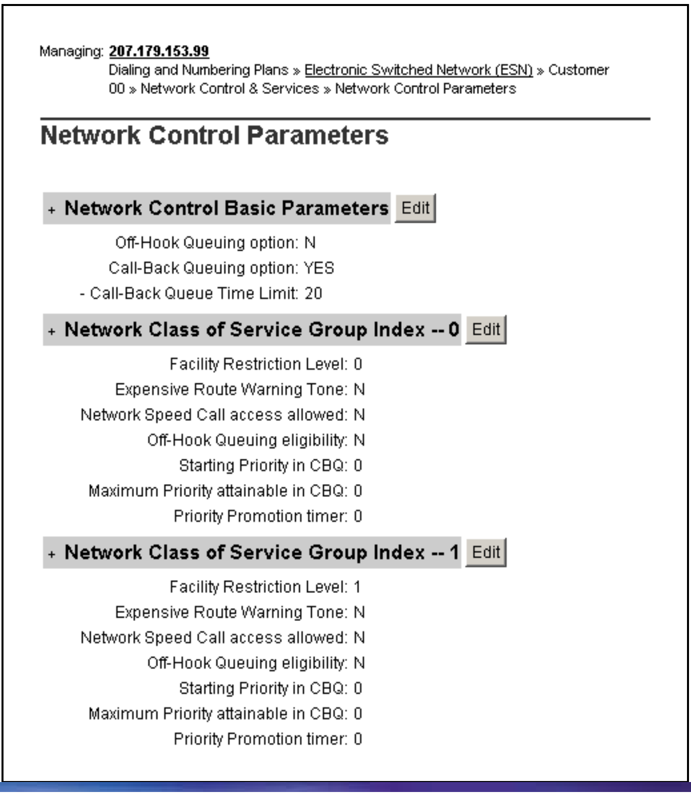
The default parameters for Network Control must be turned on.

1 Select **Dialing and Numbering Plans > Electronic Switched Network** from the EM Navigator.

2 On the **Electronic Switched Network (ESN)** web page shown in Figure 100 "Electronic Switched Network (ESN) web page" (page 199), select **Customer xx > Network Control & Service > Network Control Parameters (NCTL)**.

The **Network Control Parameters** web page opens, as shown in Figure 103 "Network Control Parameters web page" (page 201).

Figure 103
Network Control Parameters web page



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Network Control Parameters

Network Control Parameters

+ Network Control Basic Parameters

Off-Hook Queuing option: N
Call-Back Queuing option: YES
- Call-Back Queue Time Limit: 20

+ Network Class of Service Group Index -- 0

Facility Restriction Level: 0
Expensive Route Warning Tone: N
Network Speed Call access allowed: N
Off-Hook Queuing eligibility: N
Starting Priority in CBQ: 0
Maximum Priority attainable in CBQ: 0
Priority Promotion timer: 0

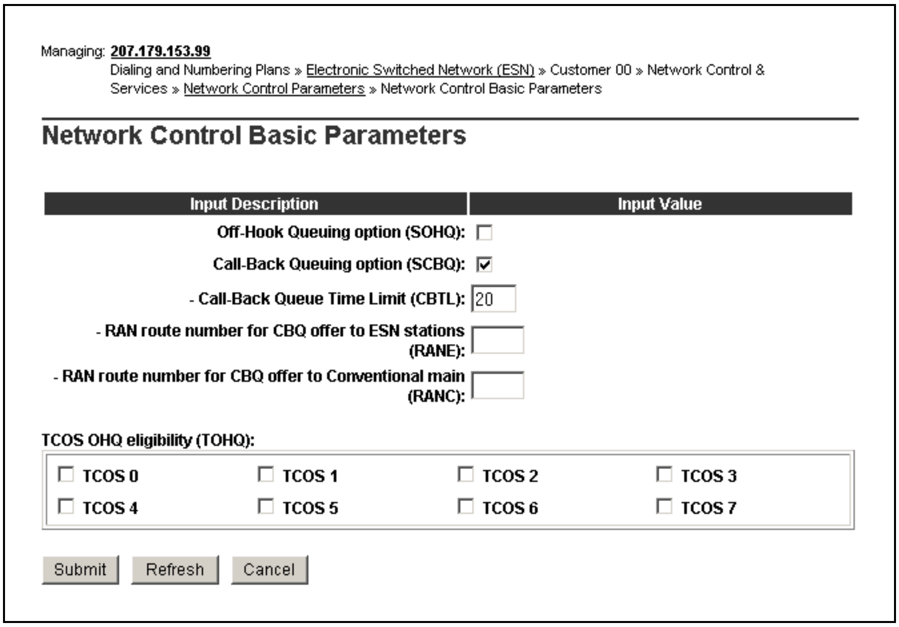
+ Network Class of Service Group Index -- 1

Facility Restriction Level: 1
Expensive Route Warning Tone: N
Network Speed Call access allowed: N
Off-Hook Queuing eligibility: N
Starting Priority in CBQ: 0
Maximum Priority attainable in CBQ: 0
Priority Promotion timer: 0

3 Click **Edit** to the right of **Network Control Basic Parameters**.

The **Network Control Basic Parameters** web page opens, as shown in Figure 104 "Network Control Basic Parameters" (page 202).

Figure 104
Network Control Basic Parameters



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Network Control Parameters » Network Control Basic Parameters

Network Control Basic Parameters

Input Description	Input Value
Off-Hook Queuing option (SOHQ):	□
Call-Back Queuing option (SCBQ):	☑
- Call-Back Queue Time Limit (CBTL):	20
- RAN route number for CBQ offer to ESN stations (RANE):	
- RAN route number for CBQ offer to Conventional main (RANC):	

TCOS OHQ eligibility (TOHQ):

□ TCOS 0	□ TCOS 1	□ TCOS 2	□ TCOS 3
□ TCOS 4	□ TCOS 5	□ TCOS 6	□ TCOS 7

4 Click **Submit** to accept the default parameters on the **Network Control Basic Parameters** web page.

The **Network Control Parameters** web page reopens.

—End—

Configuring call routing

Procedure 18 "Configuring digit manipulation tables" (page 223) must be performed before Configuring the Route List Block.

Procedure 13
Configuring the Route List Block

Step Action

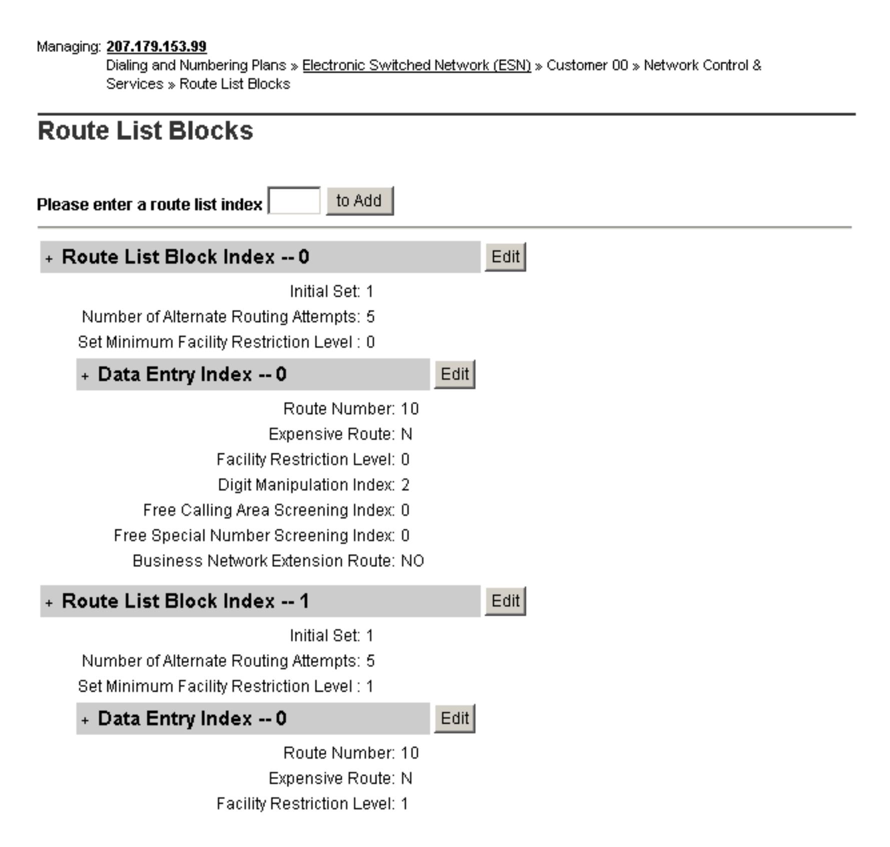
This procedure creates the Route List Block that routes calls over the Virtual Trunk route.

1 Select **Dialing and Numbering Plans > Electronic Switched Network** from the EM Navigator.

2 On the **Electronic Switched Network (ESN)** web page shown in Figure 100 "Electronic Switched Network (ESN) web page" (page 199), select **Customer xx > Network Control & Service > Route List Block (RLB)**.

The **Route List Blocks** web page opens, as shown in Figure 105 "Route List Blocks web page" (page 203).

Figure 105
Route List Blocks web page



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Route List Blocks

Route List Blocks

Please enter a route list index [ ] to Add

+ Route List Block Index -- 0 Edit

Initial Set: 1
Number of Alternate Routing Attempts: 5
Set Minimum Facility Restriction Level : 0

+ Data Entry Index -- 0 Edit

Route Number: 10
Expensive Route: N
Facility Restriction Level: 0
Digit Manipulation Index: 2
Free Calling Area Screening Index: 0
Free Special Number Screening Index: 0
Business Network Extension Route: NO

+ Route List Block Index -- 1 Edit

Initial Set: 1
Number of Alternate Routing Attempts: 5
Set Minimum Facility Restriction Level : 1

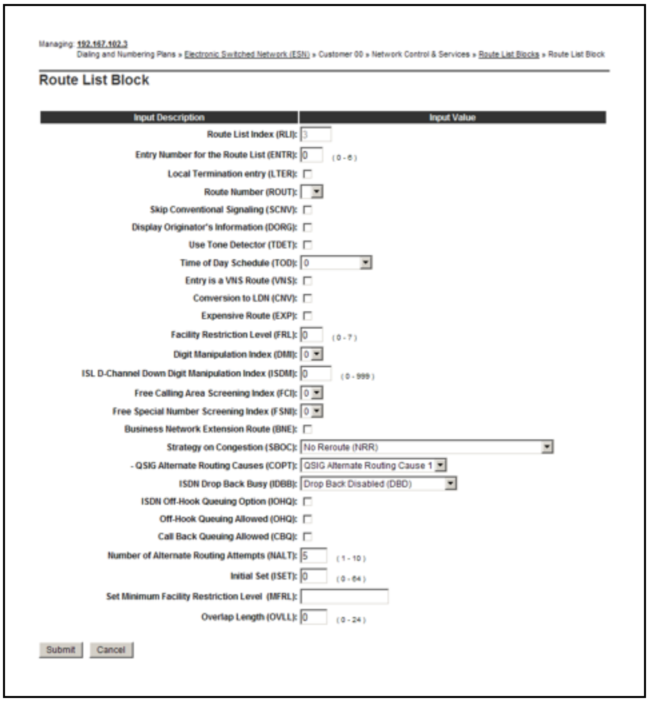
+ Data Entry Index -- 0 Edit

Route Number: 10
Expensive Route: N
Facility Restriction Level: 1

3 Enter the route list index number in the **Please enter a route list index** text box and click **to Add**.

The **Route List Block** web page opens, as shown in Figure 106 "Route List Block" (page 204).

Figure 106
Route List Block



Managing: 192.167.102.3
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Route List Blocks » Route List Block

Route List Block

Input Description	Input Value
Route List Index (RLI):	3
Entry Number for the Route List (ENTR):	0 (0 - 6)
Local Termination entry (LTER):	□
Route Number (ROUT):	▾
Skip Conventional Signaling (SCNV):	□
Display Originator's Information (DORG):	□
Use Tone Detector (TDET):	□
Time of Day Schedule (TOD):	0 ▾
Entry is a VNS Route (VNS):	□
Conversion to LDN (CNV):	□
Expensive Route (EXP):	□
Facility Restriction Level (FRL):	0 (0 - 7)
Digit Manipulation Index (DMI):	0 ▾
ISL D-Channel Down Digit Manipulation Index (ISDM):	0 (0 - 999)
Free Calling Area Screening Index (FCI):	0 ▾
Free Special Number Screening Index (FSNI):	0 ▾
Business Network Extension Route (BNE):	□
Strategy on Congestion (SBOC):	No Reroute (NRR) ▾
- QSIG Alternate Routing Causes (COPT):	QSIG Alternate Routing Cause 1 ▾
ISDN Drop Back Busy (IDBB):	Drop Back Disabled (DBD) ▾
ISDN Off-Hook Queuing Option (IOHQ):	□
Off-Hook Queuing Allowed (OHQ):	□
Call Back Queuing Allowed (CBQ):	□
Number of Alternate Routing Attempts (NALT):	5 (1 - 10)
Initial Set (ISET):	0 (0 - 64)
Set Minimum Facility Restriction Level (MFRL):	
Overlap Length (OVLL):	0 (0 - 24)

4 Fill in the appropriate information and click **Submit**.

The new Route List Block is generated, and the initial **Route List Blocks** web page reopens.

—End—

Procedure 14
Configuring Steering Codes

Step	Action

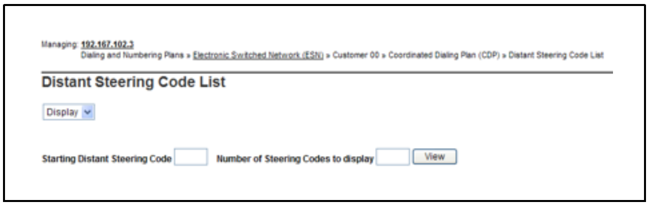
This procedure defines how digits for a call are routed under a Coordinated Dialing Plan.

1 Select **Dialing and Numbering Plans > Electronic Switched Network** from the EM Navigator.

2 On the **Electronic Switched Network (ESN)** web page shown in Figure 100 "Electronic Switched Network (ESN) web page" (page 199), select **Customer xx > Coordinated Dialing Plan (CDP) > Distant Steering Code (DSC)**.

The **Distant Steering Code List** web page opens, as shown in Figure 107 "EM Distant Steering Code List web page" (page 205).

Figure 107
EM Distant Steering Code List web page



Managing: 192.167.102.3
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Coordinated Dialing Plan (CDP) » Distant Steering Code List

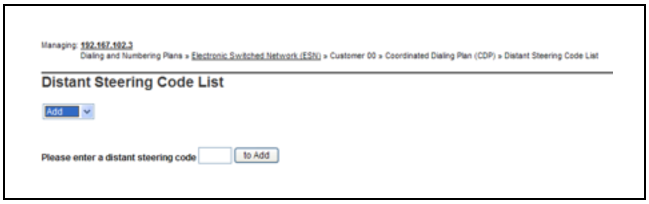
Distant Steering Code List

Display

Starting Distant Steering Code ___ Number of Steering Codes to display ___ View

Select **Add** from the drop-down list. The **Distant Steering Code List** web page refreshes, as shown in Figure 108 "EM Distant Steering Code List web page refreshed" (page 205).

Figure 108
EM Distant Steering Code List web page refreshed



Managing: 192.167.102.3
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Coordinated Dialing Plan (CDP) » Distant Steering Code List

Distant Steering Code List

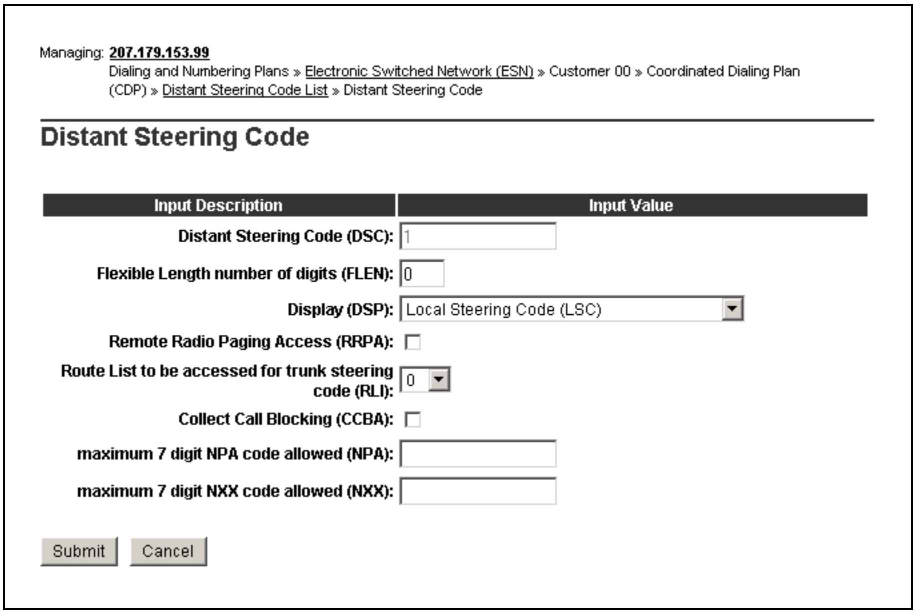
Add

Please enter a distant steering code ___ to Add

3 Enter the steering code in the **Please enter a distant steering code** text box and click **to Add**.

The **Distant Steering Code** web page opens, as shown in Figure 109 "Distant Steering Code web page" (page 205).

Figure 109
Distant Steering Code web page



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Coordinated Dialing Plan (CDP) » Distant Steering Code List » Distant Steering Code

Distant Steering Code

Input Description	Input Value
Distant Steering Code (DSC):	1
Flexible Length number of digits (FLEN):	0
Display (DSP):	Local Steering Code (LSC)
Remote Radio Paging Access (RRPA):	□
Route List to be accessed for trunk steering code (RLI):	0
Collect Call Blocking (CCBA):	□
maximum 7 digit NPA code allowed (NPA):	
maximum 7 digit NXX code allowed (NXX):	

Submit Cancel

4 Fill in the appropriate information and click **Submit**.

The **Distant Steering Code List** web page reopens.

—End—

Configuring codecs

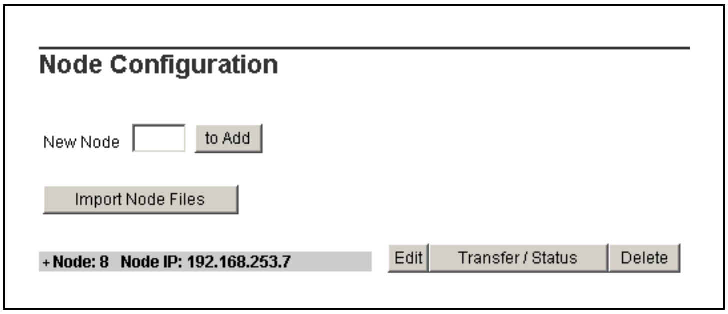
Procedure 15
Configuring codecs

Step Action

1 Select **System > IP Network > Nodes: Servers, Media Cards** from the EM Navigator.

The **Node Configuration** web page opens, as shown in Figure 110 "Node Configuration web page" (page 206).

Figure 110
Node Configuration web page



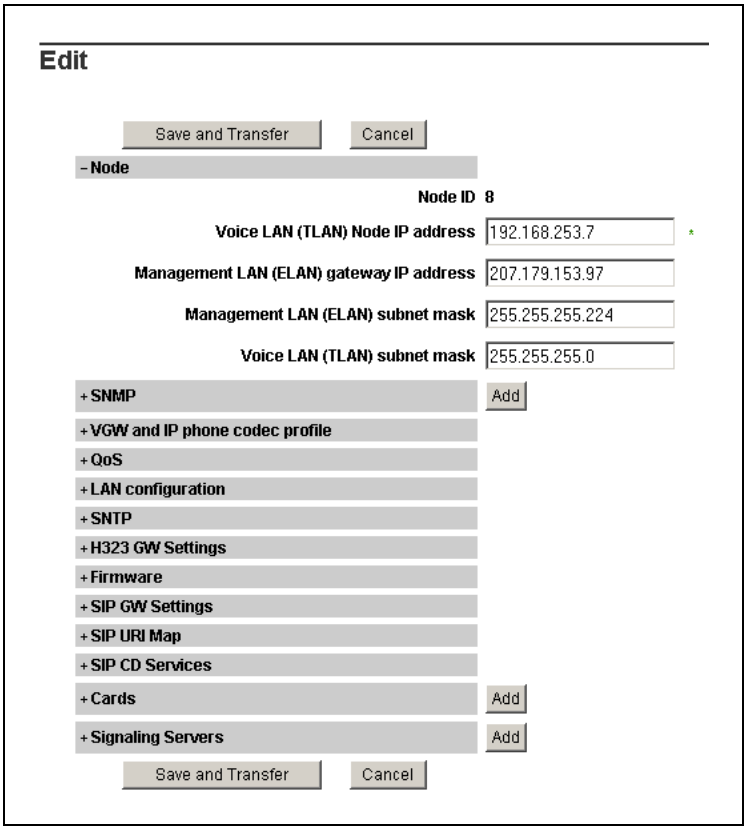
Node Configuration

New Node

2 Click **Edit** for the appropriate node.

The **Edit** web page opens, as shown in Figure 111 "Edit web page" (page 207).

Figure 111
Edit web page



Node ID 8

Voice LAN (TLAN) Node IP address 192.168.253.7 *

Management LAN (ELAN) gateway IP address 207.179.153.97

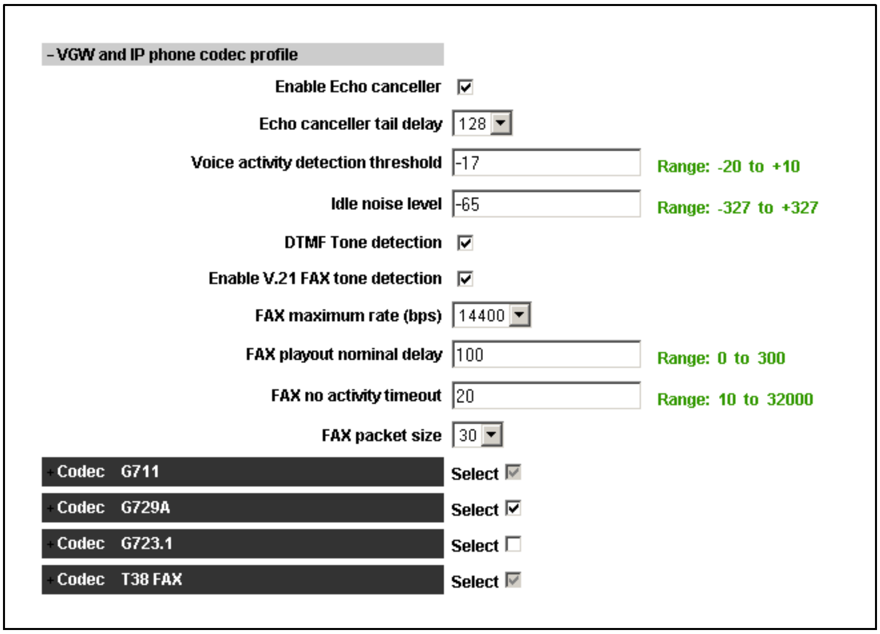
Management LAN (ELAN) subnet mask 255.255.255.224

Voice LAN (TLAN) subnet mask 255.255.255.0

3 Click on **VGW and IP phone codec profile** to open the parameter list as shown in Figure 112 "VGW and IP Phone codec profile" (page 208).

This area also includes a list of codecs.

Figure 112
VGW and IP Phone codec profile



- VGW and IP phone codec profile

Enable Echo canceller ☑

Echo canceller tail delay 128

Voice activity detection threshold -17 Range: -20 to +10

Idle noise level -65 Range: -327 to +327

DTMF Tone detection ☑

Enable V.21 FAX tone detection ☑

FAX maximum rate (bps) 14400

FAX playout nominal delay 100 Range: 0 to 300

FAX no activity timeout 20 Range: 10 to 32000

FAX packet size 30

+ Codec G711 Select ☑

+ Codec G729A Select ☑

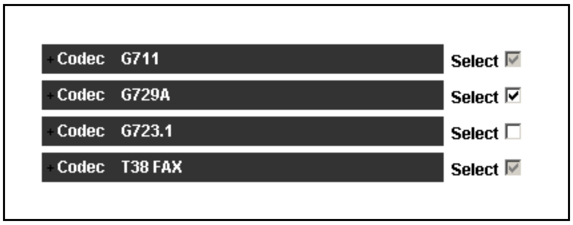
+ Codec G723.1 Select ☐

+ Codec T38 FAX Select ☑

4 To configure a codec, select the **Select** check box to the right of the codec name. For example, in Figure 113 "Example of a selected codec — G.729A" (page 208) the G.729A codec has been selected.

> ***Note:*** The G.711 and T38 FAX codecs are automatically selected.

Figure 113
Example of a selected codec G.729A



+ Codec G711 Select ☑

+ Codec G729A Select ☑

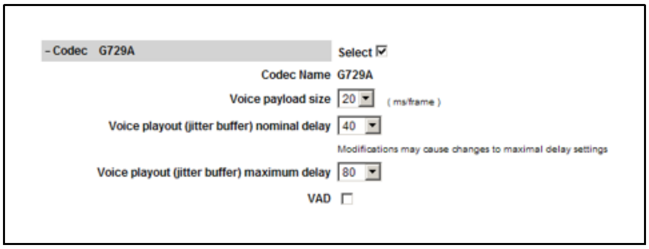
+ Codec G723.1 Select ☐

+ Codec T38 FAX Select ☑

5 Click on the codec name to modify **the Voice payload size (ms/frame), Voice playout (jitter buffer) nominal delay**, and **Voice playout (jitter buffer) maximum delay** values of a codec.

Use the drop-down lists to choose the values. See the example in Figure 114 "Example of G.729A settings" (page 209).

Figure 114
Example of G.729A settings



- Codec G729A

Select ☑

Codec Name G729A

Voice payload size 20 (ms/frame)

Voice playout (jitter buffer) nominal delay 40

Modifications may cause changes to maximal delay settings

Voice playout (jitter buffer) maximum delay 80

VAD □

6 Repeat step 4 and step 5 for each codec that requires configuration.

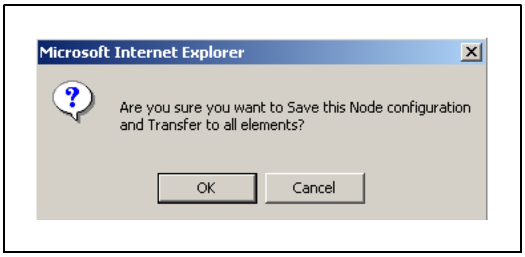
> ***Note:*** For detailed information about configuring codecs, refer to *Converging the Data Network with VoIP* (NN43001-260) and *IP Line: Description, Installation, and Operation* (NN43100-500).

7 Click **Save and Transfer**.

This saves the changes and transfers the node configuration files to all elements in the node (that is, Signaling Servers, Call Server, and Voice Gateway Media Cards).

A warning dialog box opens asking if you want to save and transfer the configuration changes (see Figure 115 "Save and Transfer dialog box" (page 209)).

Figure 115
Save and Transfer dialog box



Microsoft Internet Explorer

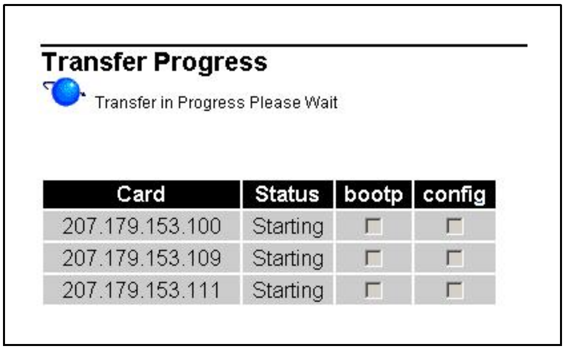
Are you sure you want to Save this Node configuration and Transfer to all elements?

8 Click **OK**.

A series of pages including the following display:

- **Transfer Progress** web page (see Figure 116 "Transfer Progress — Starting" (page 210) and Figure 117 "Transfer Progress — Transferring" (page 210), and Figure 118 "Transfer Progress — Completed" (page 210))
- **Transfer Failure Report** web page (if applicable)
- **Transfer / Status** web page (see Figure 119 "Transfer / Status" (page 211)) This web page shows if the transfer was successful, and allows the node information to be transferred again.

Figure 116
Transfer Progress Starting

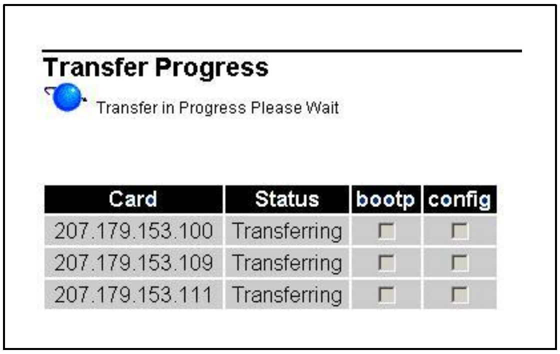


Transfer Progress

Transfer in Progress Please Wait

Card	Status	bootp	config
207.179.153.100	Starting	□	□
207.179.153.109	Starting	□	□
207.179.153.111	Starting	□	□

Figure 117
Transfer Progress Transferring

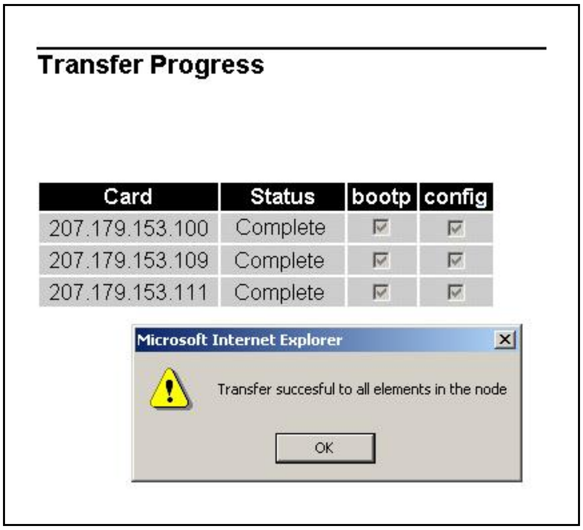


Transfer Progress

Transfer in Progress Please Wait

Card	Status	bootp	config
207.179.153.100	Transferring	□	□
207.179.153.109	Transferring	□	□
207.179.153.111	Transferring	□	□

Figure 118
Transfer Progress Completed



Transfer Progress

Card	Status	bootp	config
207.179.153.100	Complete	☑	☑
207.179.153.109	Complete	☑	☑
207.179.153.111	Complete	☑	☑

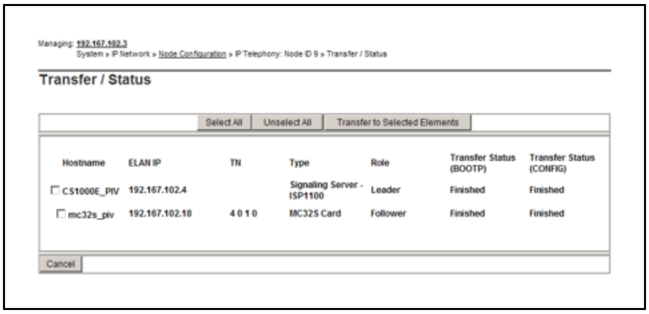
Microsoft Internet Explorer

Transfer succesful to all elements in the node

9 Click **OK**.

The **Transfer / Status** web page opens, as shown in Figure 119 "Transfer / Status" (page 211). This **Transfer / Status** web page allows you to transfer the configuration to selected elements or all elements.

Figure 119
Transfer / Status



Managing: 192.167.102.3
System » IP Network » Node Configuration » IP Telephony: Node ID 9 » Transfer / Status

Transfer / Status

Hostname	ELAN IP	TN	Type	Role	Transfer Status (BOOTP)	Transfer Status (CONFIG)
☐ CS1000E_PIV	192.167.102.4		Signaling Server - ISP1100	Leader	Finished	Finished
☐ mc32s_piv	192.167.102.18	4 0 1 0	MC32S Card	Follower	Finished	Finished

10 Click **Cancel** to return to the **Node Configuration** web page.

The **Node Configuration** web page (Figure 110 "Node Configuration web page" (page 206)) reopens.

> ***Note:*** When on the **Node Configuration** web page, clicking **Transfer / Status** opens the **Transfer / Status** web page (see Figure 119 "Transfer / Status" (page 211)). This page is used to send the node configuration files to all IP Telephony components in the node.

- If any element within the Node fails to transfer either BOOTP or CONFIG files, the **Transfer / Status** button is highlighted in red.
- The **Transfer / Status** button is highlighted in yellow if the transfer status of the node elements is unavailable.

—End—

Configuring QoS (DiffServ) values

Quality of Service (QoS) values are configured through Element Manager.

Procedure 16
Configuring QoS (DiffServ) values

Step **Action**

1 Select **System > IP Network > Nodes: Servers, Media Cards** from the EM Navigator.

The **Node Configuration** web page opens, as shown in Figure 110 "Node Configuration web page" (page 206).

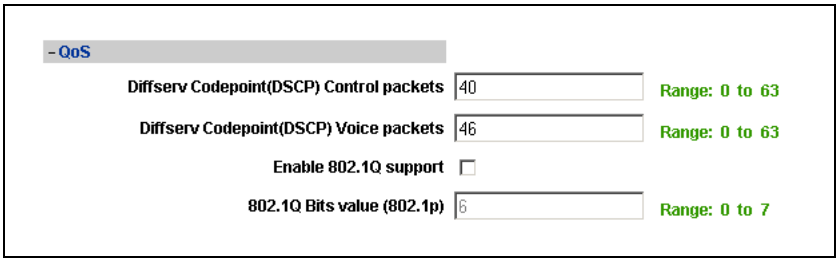
2 Click **Edit** for the appropriate node.

The **Edit** web page opens, as shown in Figure 111 "Edit web page" (page 207).

3 Click **QoS**.

The QoS section expands, as shown in Figure 120 "QoS section" (page 212).

Figure 120
QoS section



- QoS

Diffserv Codepoint(DSCP) Control packets 40 Range: 0 to 63

Diffserv Codepoint(DSCP) Voice packets 46 Range: 0 to 63

Enable 802.1Q support □

802.1Q Bits value (802.1p) 6 Range: 0 to 7

4 Enter the recommended values:

a. **Diffserv Codepoint (DSCP) Control packets =** 40 - Class Selector 5 (CS5). The range is 0 – 63. This configures the priority of the signaling messaging.

b. **Diffserv CodePoint (DSCP) Voice packets =** 46 - Control DSCP - Expedited Forwarding (EF). The range is 0 – 63.

Note: The Differentiated Service (DiffServ) CodePoint (DSCP) determines the priorities of the management and voice packets in the IP Line network. The values are stored in IP telephony CONFIG.INI file. The values used in the IP packets are respectively **160** (40*4) and **184** (46*4).

5 Click **Save and Transfer**.

For more information about Differentiated Service (DiffServ) CodePoint (DSCP), see *Converging the Data Network with VoIP* (NN43001-260) and *IP Line: Description, Installation, and Operation* (NN43100-500).

—End—

Configuring call types

To configure call types and location codes HLOC, HNPA, LOC, NPA, NXX, SPN using Element Manager, follow the steps in Procedure 17 "Configuring call types" (page 213).

Procedure 17
Configuring call types

Step	Action

1 Select **Dialing and Numbering Plans > Electronic Switched Network** from the EM Navigator.

The **Electronic Switched Network (ESN)** web page opens.

2 Scroll to the **Numbering Plan (NET)** link (see Figure 121 "Numbering Plan (NET)" (page 214)).

To configure...	See...
Home Location Code (HLOC)	step 3
Home Area Code (HNPA)	step 4
Location Code (LOC)	step 5
Numbering Plan Area Code (NPA)	step 6
Exchange (Central Office) Code (NXX)	step 7
Special Number (SPN)	step 8

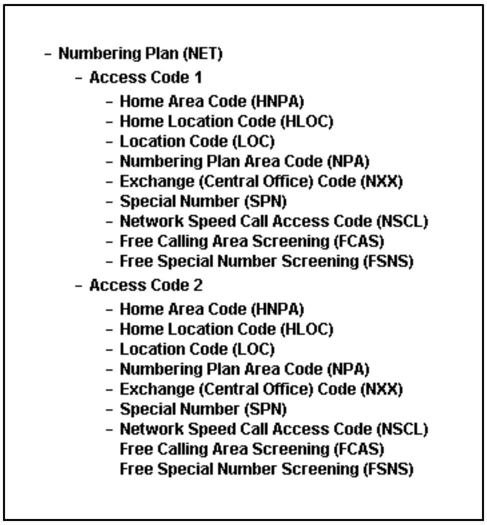
Note 1: Do not provision non-North American numbers as NPA or NXX if you want to configure overlap signaling, as these are still 100% en bloc. For more information about overlap signaling, refer to "Overlap signaling" (page 251).

Note 2: If you use the SPN to provide NPA and NXX equivalents, these can remain associated with the two ESN access codes (that is, AC1 = 6 and AC2 = 9).

- If the destination is accessed by way of another CS 1000 system, then leave the number as an SPN and translate it at the interface to the PSTN.
- If the destination is accessed by way of any other device, then perform call-type conversion as required for that device. Usually, this means changing the call type to national or subscriber (NPA, NXX) in the DMI of the Call Server sending out the number. (Overlap signaling allows this use of NPA and NXX, since the call began as an SPN. This allows national and local number overlap to a third party.)

Note 3: To get an HNPA equivalent with SPN, use local termination (LTER) in the RLI and delete the prefix.

Figure 121
Numbering Plan (NET)



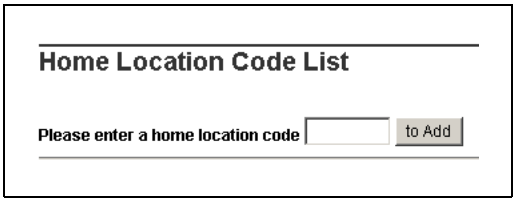
```
- Numbering Plan (NET)
   - Access Code 1
      - Home Area Code (HNPA)
      - Home Location Code (HLOC)
      - Location Code (LOC)
      - Numbering Plan Area Code (NPA)
      - Exchange (Central Office) Code (NXX)
      - Special Number (SPN)
      - Network Speed Call Access Code (NSCL)
      - Free Calling Area Screening (FCAS)
      - Free Special Number Screening (FSNS)
   - Access Code 2
      - Home Area Code (HNPA)
      - Home Location Code (HLOC)
      - Location Code (LOC)
      - Numbering Plan Area Code (NPA)
      - Exchange (Central Office) Code (NXX)
      - Special Number (SPN)
      - Network Speed Call Access Code (NSCL)
        Free Calling Area Screening (FCAS)
        Free Special Number Screening (FSNS)
```

3 To configure Home Location Code, perform the following steps:

a. Click **Home Location Code (HLOC)** under **Access Code 1** or **Access Code 2**.

The **Home Location Code List** web page opens, as shown in Figure 122 "Home Location Code List web page" (page 214).

Figure 122
Home Location Code List web page



Home Location Code List

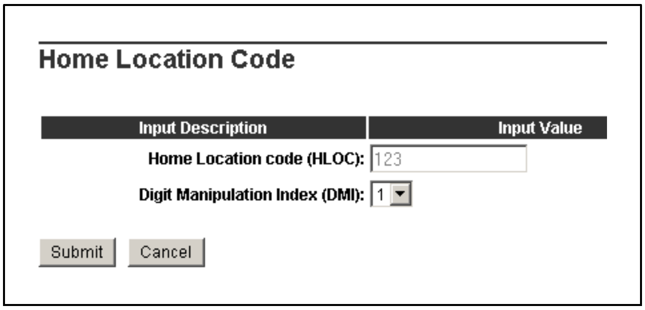
Please enter a home location code

b. Enter a code in the **home location code** text box.

c. Click **to Add**.

The **Home Location Code** web page opens, as shown in Figure 123 "Home Location Code web page" (page 215). The **Home Location code (HLOC)** is auto-filled.

Figure 123
Home Location Code web page


Home Location Code

Input Description	Input Value
Home Location code (HLOC):	123
Digit Manipulation Index (DMI):	1

d. Select a **Digit Manipulation Index (DMI)**.

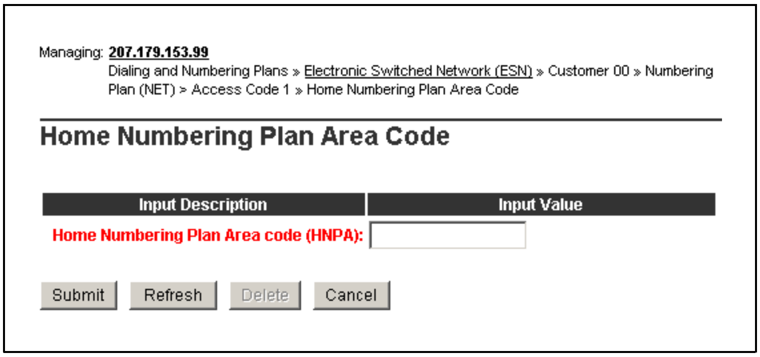
e. Click **Submit**.

4 To configure Home Area Code (HNPA), perform the following steps:

a. Click **Home Area Code (HNPA)** under **Access Code 1** or **Access Code 2**

The **Home Numbering Plan Area Code** web page opens, as shown in Figure 124 "Home Numbering Plan Area Code web page" (page 215).

Figure 124
Home Numbering Plan Area Code web page


Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Numbering Plan (NET) » Access Code 1 » Home Numbering Plan Area Code

Home Numbering Plan Area Code

Input Description	Input Value
Home Numbering Plan Area code (HNPA):	

b. Enter the **Home Number Plan Area code (HNPA)** in the text box.

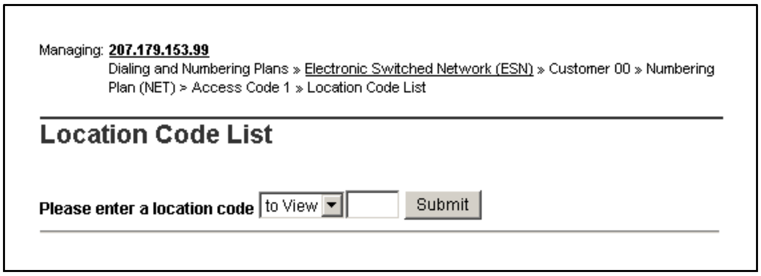
c. Click **Submit**.

5 To configure Location Code (LOC), perform the following steps:

a. Click **Location Code (LOC)** under **Access Code 1** or **Access Code 2**.

The **Location Code List** web page opens, as shown in Figure 125 "Location Code List web page" (page 216).

Figure 125
Location Code List web page



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Numbering Plan (NET) » Access Code 1 » Location Code List

Location Code List

Please enter a location code: to View

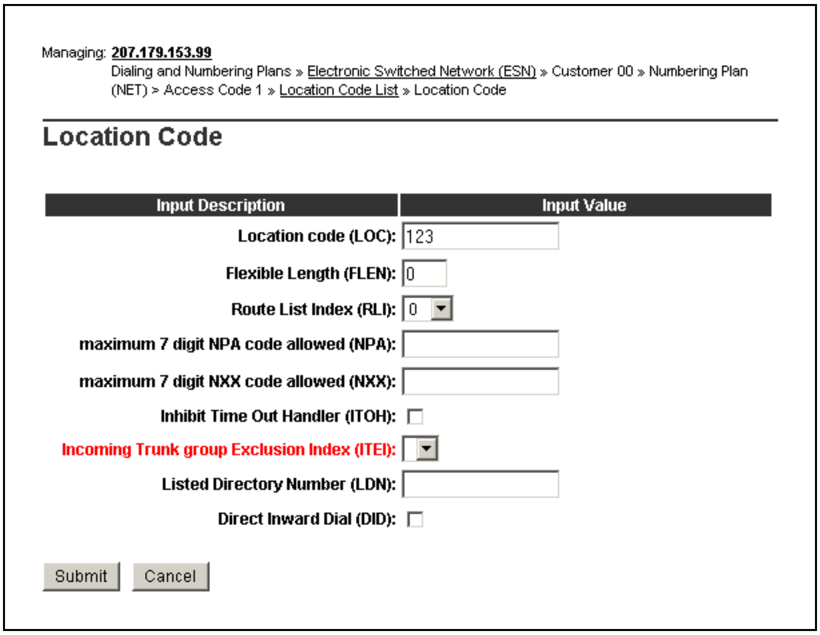
b. Select **to Add** from the drop-down list.

c. Enter a code in the **location code** text box.

d. Click **Submit**.

The **Location Code** web page opens, as shown in Figure 126 "Location Code web page" (page 216).

Figure 126
Location Code web page



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Numbering Plan (NET) » Access Code 1 » Location Code List » Location Code

Location Code

Input Description	Input Value
Location code (LOC):	123
Flexible Length (FLEN):	0
Route List Index (RLI):	0
maximum 7 digit NPA code allowed (NPA):	
maximum 7 digit NXX code allowed (NXX):	
Inhibit Time Out Handler (ITOH):	□
Incoming Trunk group Exclusion Index (ITEI):	
Listed Directory Number (LDN):	
Direct Inward Dial (DID):	□

e. Enter the appropriate information.

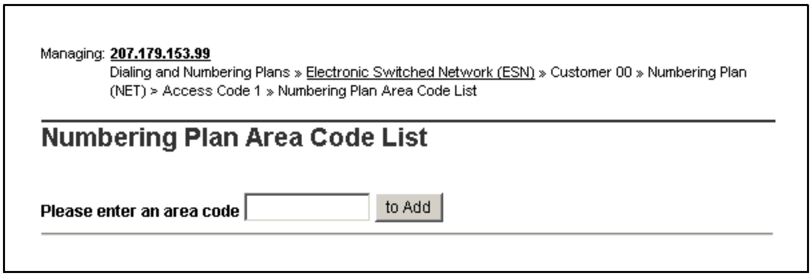
f. Click **Submit**.

6 To configure Number Plan Area Code (NPA), perform the following steps:

a. Click **Numbering Plan Area Code (NPA)** under **Access Code 1** or **Access Code 2**.

The **Numbering Plan Area Code List** web page opens, as shown in Figure 127 "Numbering Plan Area Code List web page" (page 217).

Figure 127
Numbering Plan Area Code List web page



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Numbering Plan (NET) » Access Code 1 » Numbering Plan Area Code List

Numbering Plan Area Code List

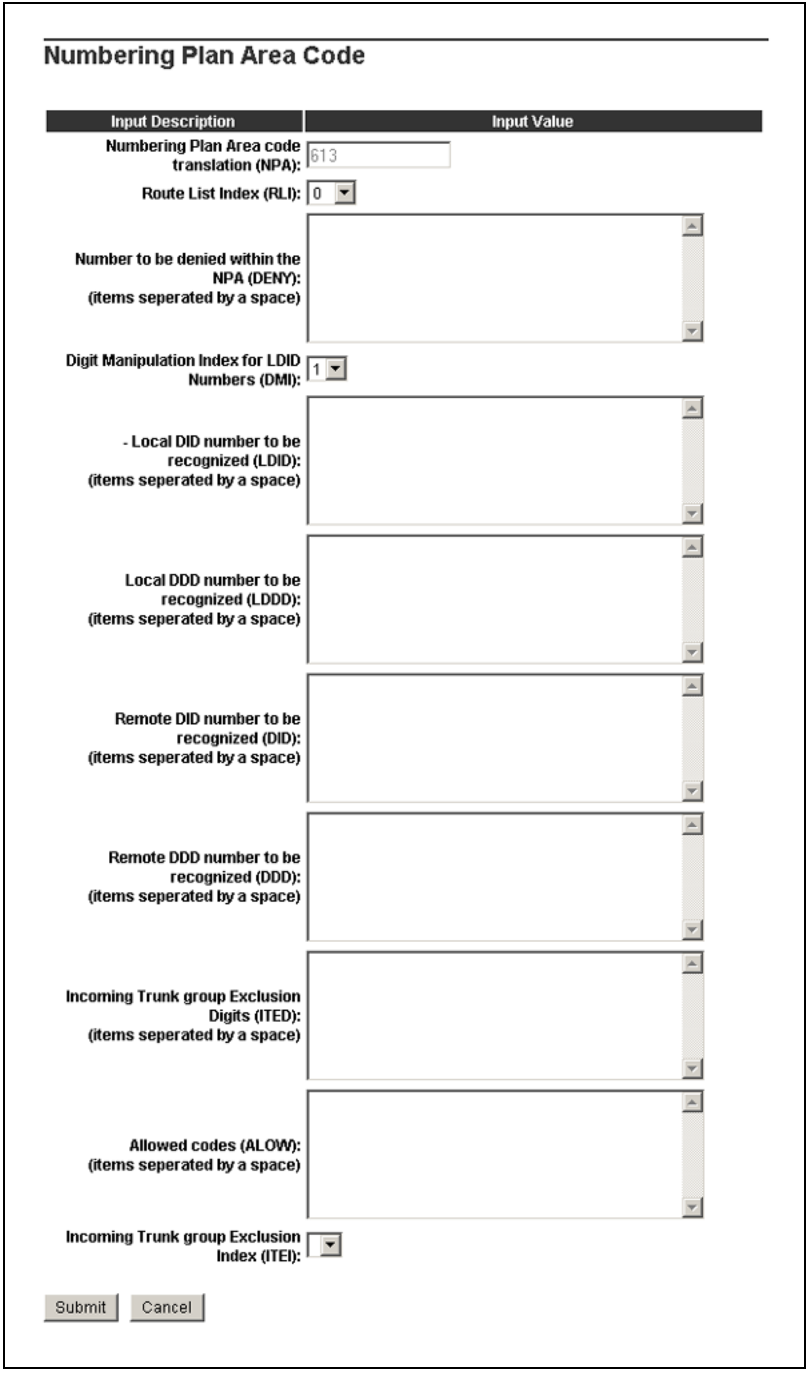
Please enter an area code

b. Enter an area code.

c. Click **to Add**.

The **Numbering Plan Area Code** web page opens, as shown in Figure 128 "Numbering Plan Area Code web page" (page 218).

Figure 128
Numbering Plan Area Code web page



Numbering Plan Area Code

Input Description	Input Value
Numbering Plan Area code translation (NPA):	613
Route List Index (RLI):	0
Number to be denied within the NPA (DENY): (items seperated by a space)	
Digit Manipulation Index for LDID Numbers (DMI):	1
- Local DID number to be recognized (LDID): (items seperated by a space)	
Local DDD number to be recognized (LDDD): (items seperated by a space)	
Remote DID number to be recognized (DID): (items seperated by a space)	
Remote DDD number to be recognized (DDD): (items seperated by a space)	
Incoming Trunk group Exclusion Digits (ITED): (items seperated by a space)	
Allowed codes (ALOW): (items seperated by a space)	
Incoming Trunk group Exclusion Index (ITEI):	

d. Enter the appropriate information.

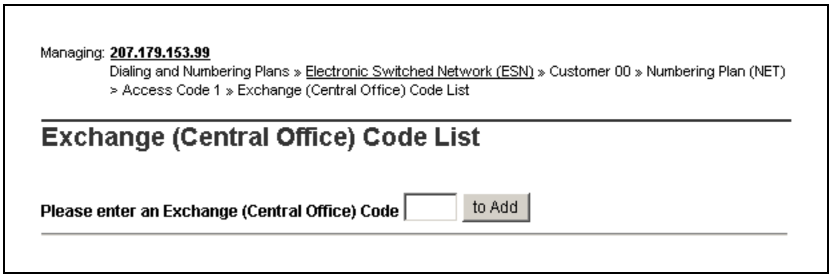
e. Click **Submit**.

7 To configure Exchange (Central Office) Code (NXX), perform the following steps:

a. Click **Exchange (Central Office) Code (NXX)** under **Access Code 1** or **Access Code 2**.

The **Exchange (Central Office) Code List** web page opens, as shown in Figure 129 "Exchange (Central Office) Code List web page" (page 219).

Figure 129
Exchange (Central Office) Code List web page



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Numbering Plan (NET) » Access Code 1 » Exchange (Central Office) Code List

Exchange (Central Office) Code List

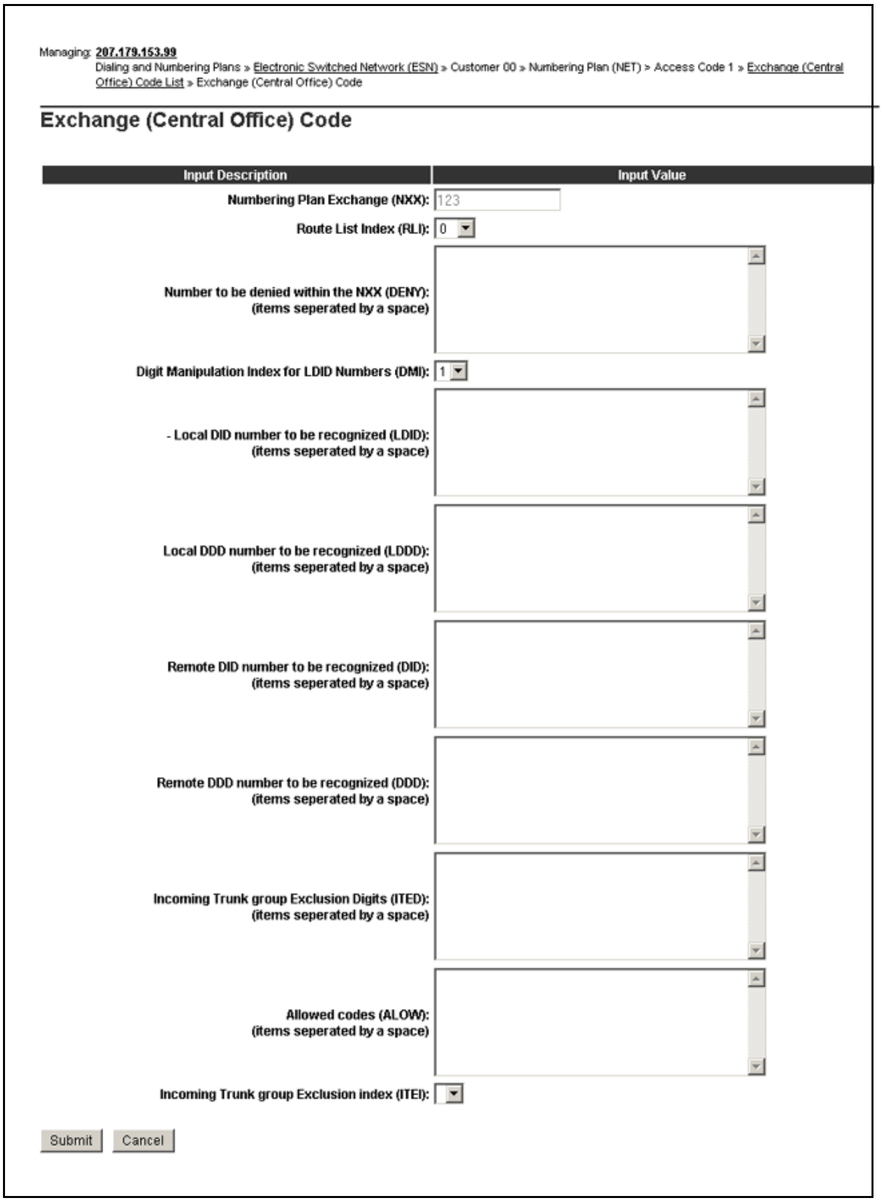
Please enter an Exchange (Central Office) Code

b. Enter the **Exchange (Central Office) Code** in the text box.

c. Click **to Add**.

The **Exchange (Central Office) Code** web page opens, as shown in Figure 130 "Exchange (Central Office) Code web page" (page 220).

Figure 130
Exchange (Central Office) Code web page



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Numbering Plan (NET) > Access Code 1 » Exchange (Central Office) Code List » Exchange (Central Office) Code

Exchange (Central Office) Code

Input Description	Input Value
Numbering Plan Exchange (NXX):	123
Route List Index (RLI):	0
Number to be denied within the NXX (DENY): (items seperated by a space)	
Digit Manipulation Index for LDID Numbers (DMI):	1
- Local DID number to be recognized (LDID): (items seperated by a space)	
Local DDD number to be recognized (LDDD): (items seperated by a space)	
Remote DID number to be recognized (DID): (items seperated by a space)	
Remote DDD number to be recognized (DDD): (items seperated by a space)	
Incoming Trunk group Exclusion Digits (ITED): (items seperated by a space)	
Allowed codes (ALOW): (items seperated by a space)	
Incoming Trunk group Exclusion index (ITEI):	

d. Enter the appropriate information.

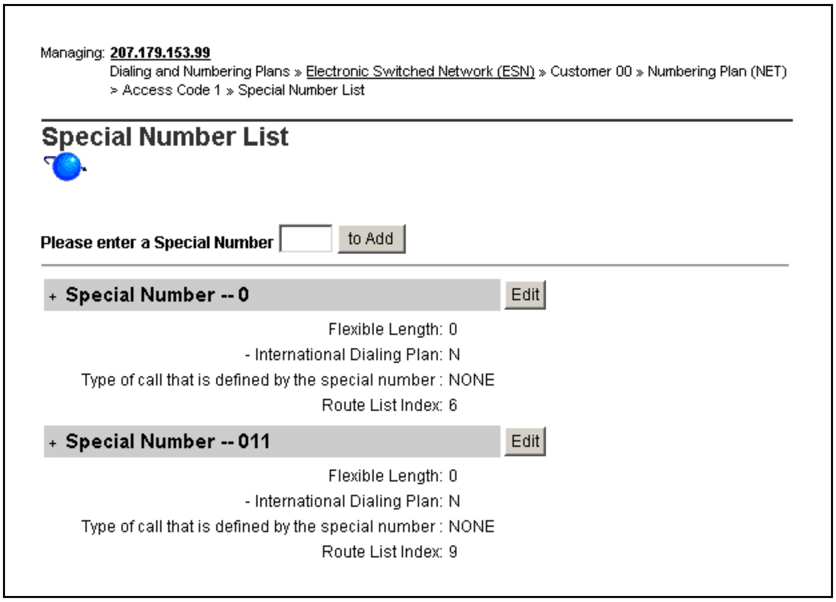
e. Click **Submit**.

8 To configure Special Number (SPN), perform the following steps:

a. Click **Special Number (SPN)** under **Access Code 1** or **Access Code 2**.

The **Special Number List** web page opens, as shown in Figure 131 "Special Number List web page" (page 221).

Figure 131
Special Number List web page



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Numbering Plan (NET) » Access Code 1 » Special Number List

Special Number List

Please enter a Special Number [ ] to Add

+ Special Number -- 0 Edit

Flexible Length: 0
- International Dialing Plan: N
Type of call that is defined by the special number : NONE
Route List Index: 6

+ Special Number -- 011 Edit

Flexible Length: 0
- International Dialing Plan: N
Type of call that is defined by the special number : NONE
Route List Index: 9

b. Enter the number.

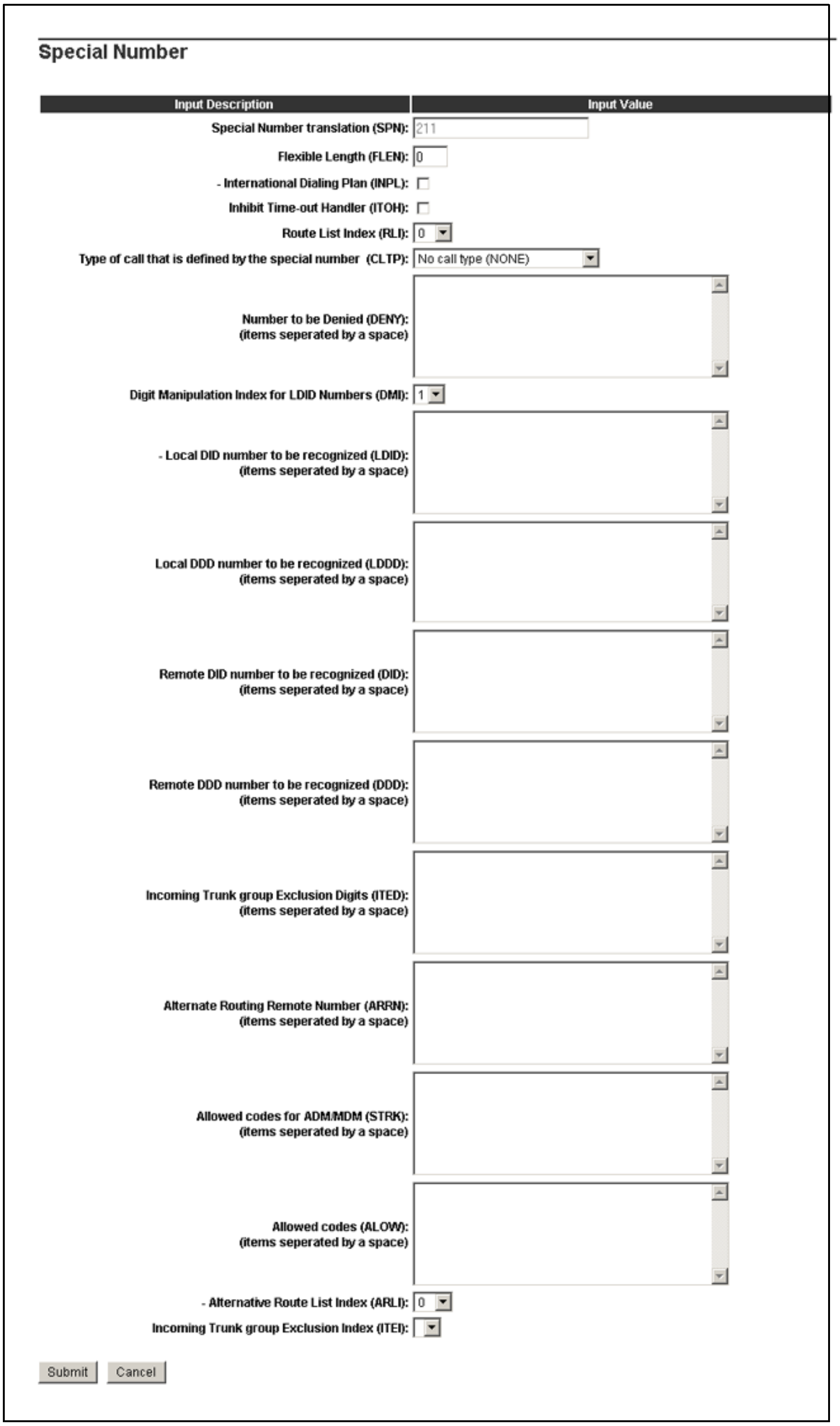
c. Click **to Add**.

The **Special Number** web page opens (see Figure 132 "Special Number" (page 222)).

d. Enter the appropriate information.

e. Click **Submit** at the bottom of the web page.

Figure 132
Special Number



Special Number

Input Description	Input Value
Special Number translation (SPN):	211
Flexible Length (FLEN):	0
- International Dialing Plan (INPL):	□
Inhibit Time-out Handler (ITOH):	□
Route List Index (RLI):	0 ▾
Type of call that is defined by the special number (CLTP):	No call type (NONE) ▾
Number to be Denied (DENY): (items seperated by a space)	
Digit Manipulation Index for LDID Numbers (DMI):	1 ▾
- Local DID number to be recognized (LDID): (items seperated by a space)	
Local DDD number to be recognized (LDDD): (items seperated by a space)	
Remote DID number to be recognized (DID): (items seperated by a space)	
Remote DDD number to be recognized (DDD): (items seperated by a space)	
Incoming Trunk group Exclusion Digits (ITED): (items seperated by a space)	
Alternate Routing Remote Number (ARRN): (items seperated by a space)	
Allowed codes for ADM/MDM (STRK): (items seperated by a space)	
Allowed codes (ALOW): (items seperated by a space)	
- Alternative Route List Index (ARLI):	0 ▾
Incoming Trunk group Exclusion Index (ITEI):	▾

f. Enter the appropriate information.

g. Click **Submit** at the bottom of the web page.

—End—

Configuring digit manipulation tables

Procedure 18
Configuring digit manipulation tables

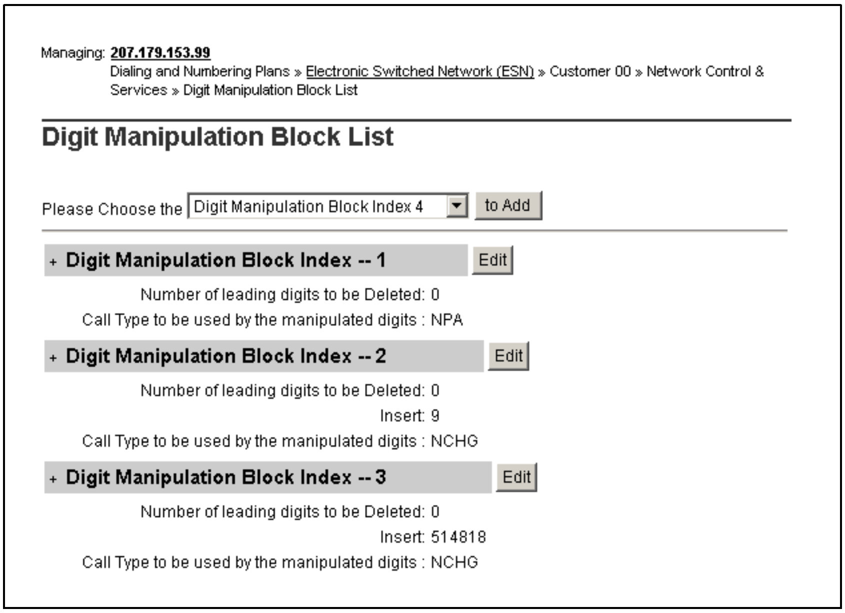
Step Action

1 Select **Dialing and Numbering Plans > Electronic Switched Network** from the EM Navigator.

2 On the **Electronic Switched Network (ESN)** web page shown in Figure 100 "Electronic Switched Network (ESN) web page" (page 199), select **Customer xx > Network Control & Services > Digit Manipulation Block (DGT)**.

The **Digit Manipulation Block List** web page opens, as shown in Figure 133 "Digit Manipulation Block List web page" (page 223).

Figure 133
Digit Manipulation Block List web page



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Digit Manipulation Block List

Digit Manipulation Block List

Please Choose the Digit Manipulation Block Index 4 to Add

+ Digit Manipulation Block Index -- 1 Edit
Number of leading digits to be Deleted: 0
Call Type to be used by the manipulated digits : NPA

+ Digit Manipulation Block Index -- 2 Edit
Number of leading digits to be Deleted: 0
Insert: 9
Call Type to be used by the manipulated digits : NCHG

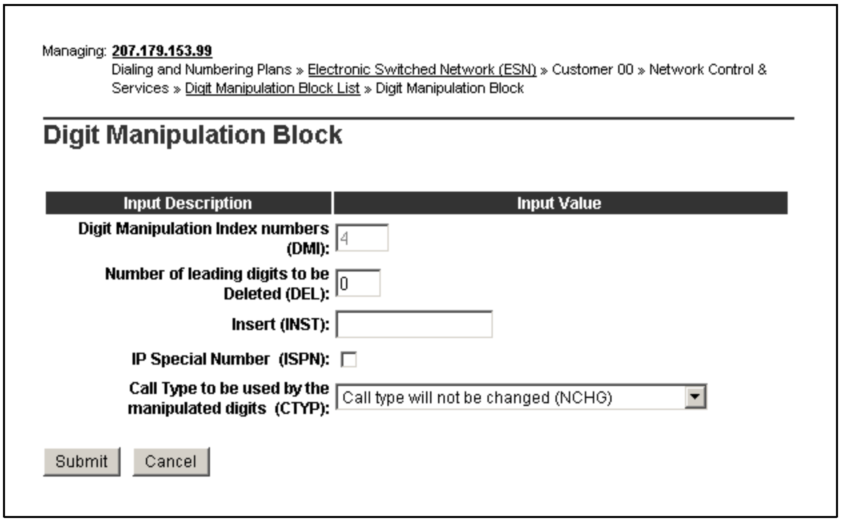
+ Digit Manipulation Block Index -- 3 Edit
Number of leading digits to be Deleted: 0
Insert: 514818
Call Type to be used by the manipulated digits : NCHG

3 Select a **Digit Manipulation Block Index** number in the drop-down list.

4 Click **to Add**.

The **Digit Manipulation Block** web page opens, as shown in Figure 134 "Digit Manipulation Block web page" (page 224).

Figure 134
Digit Manipulation Block web page



Managing: 207.179.153.99
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Digit Manipulation Block List » Digit Manipulation Block

Digit Manipulation Block

Input Description	Input Value
Digit Manipulation Index numbers (DMI):	4
Number of leading digits to be Deleted (DEL):	0
Insert (INST):	
IP Special Number (ISPN):	□
Call Type to be used by the manipulated digits (CTYP):	Call type will not be changed (NCHG)

5 Enter the appropriate information.

6 Click **Submit**.

—End—

Feature Implementation of IP Peer Networking

If you are using the Command Line Interface (CLI), use the following implementation tables to configure the IP Peer Networking feature.

Task summary list

The following is a summary of the tasks in this section:

1. LD 17 – Configure D-channels.
2. LD 15 – Configure network settings and options.
3. LD 16 – Configure the route. This route can be configured as an H.323 route or a SIP route.
   - To configure a SIP route, see "LD 16 – Configure the SIP route." (page 227).
   - To configure an H.323 route, see "LD 16 – Configure the H.323 route." (page 229).
4. LD 97 – Configure the superloop for the Virtual Trunks.
5. LD 14 – Configure Virtual Trunks.

6. LD 86 – Configure dialing plan, networking, and ESN data.
7. LD 87 – Configure network access.
8. LD 86 – Configure the Digit Manipulation Index.
9. LD 86 – Configure the Route List Block for the Virtual Trunk route.
10. LD 87 – Configure CDP steering codes.
11. LD 90 – Configure call types and Location Codes.

LD 17 Configure D-channels.

Prompt	Response	Description
REQ	CHG	Change existing data
TYPE	ADAN	Action Device And Number
- ADAN	NEW DCH xx	Action Device And Number, where xx is 0-63.
CAB_TYPE		Cabinet Type
	IP FIBR	IP Expansion Cabinet or Media Gateway Fiber Expansion Cabinet
- CTYP		Card Type
	DCIP	D-channel over IP
- DES	x...x	Designator
BANR	YES	Enable security banner printing option
- IFC	SL1	Interface type for D-channel
CO_TYPE	aaa	Central Office switch type, where aaa = (STD) or ATT
- RCVP	YES	Auto-recovery to primary D-channel option.
- - ISLM	(4000)	Integrated Services Signaling Link Maximum The maximum number of ISL trunks controlled by the D-channel. ***Note:*** ISLM prompt is hidden for D-channel on IP and is defaulted to 4000.
- OTBF	1-(32)-127	Output Request Buffers
- RLS	xx	Release ID of the switch at the far end of the D-channel

Prompt	Response	Description
- RCAP		Remote Capabilities All nodes must use same RCAP
	ND1, ND2 , ND3	ND3 ensures same level of service between MCDN and QSIG Name Display Supplementary Service
	MWI	Message Waiting Indication support over SIP using a SIP NOTIFY message rather than an MCDN message encapsulated in SIP.
		Note: MWI is also used for H.323 if a BCM is in the network.

LD 15 Configure network settings and options.

Prompt	Response	Description
REQ:	NEW CHG	Add new block Change existing data
TYPE:	NET	ISDN and ESN Networking options
CUST		Customer number
	0-99	Range for Large System and CS 1000E system
	0-31	Range for Small System, CS 1000E system, Media Gateway 1000B, and Media Gateway 1000T
...		
OPT	a...a	Options
AC2		Access Code 2
		Enter call types that use Access Code 2 as defined in LD 86, for automatic insertion of UDP access code. Multiple responses are permitted. If a numbering plan is not entered here, it is automatically defaulted to AC1.
	NPA NXX INTL SPN LOC	E.164 National number E.164 Subscriber number International number Special Number Location Code
FNP	(YES)	Enable Flexible Numbering Plan for customer
ISDN	YES	Integrated Services Digital Network
VPNI	1-16283	Virtual Private Network Identifier
- PNI	(0)-32700	Private Network Identifier
- CLID	(NO)	Do not enable Calling Line Identification option
CNTC	xx	Country code (see Note 1)

Prompt	Response	Description
NATC	xx	National access code (see Note 1)
INTC	xxx	International access code (see Note 1)

Note 1: CNTC, NATC and INTC are needed when a public call is tandemed over the Virtual Trunk.

— CNTC is the country code for the country where the switch is located. For example, CNTC = 1 for Canada.

— NATC is the national access code. For example, NATC = 1 for Canada.

— INTC is the international access code. For example, INTC = 011 for Canada.
For example, a caller who wants to reach Austria dials 6-011-61-xxxyyyzzz from endpoint A (for example, in Toronto) over the Virtual Trunk to endpoint B (for example, in the United Kingdom) which serves as a gateway to the PSTN.

The 011 is stripped off at endpoint A because the NRS does not understand it. Endpoint B would receive 61-xxxyyyzzz and compare 61 with its CNTC (= 44) and assumes that this is an international call. So, it inserts the INTC (= 00 for Europe) and sends 00-61-xxxyyyzzz to the PSTN routing to Austria.

Consider another caller from endpoint A making a call to the UK PSTN by dialing 6-011-44-xxxyyyzzz. Endpoint B would receive 44-xxxyyyzzz. It finds that 44 equals to its CNTC and figures that this is a national call. So, it strips off 44 and inserts the NATC (= 0 for UK) and sends 0-xxxyyyzzz to the PSTN.

Note 2: In the Route Data Block, the zone parameter makes the codec selections and calculates the bandwidth usage for calls to the trunk members of a given route.

Configure the routes:

- To configure a SIP route, see "LD 16 – Configure the SIP route." (page 227) below.
- To configure an H.323 route, see "LD 16 – Configure the H.323 route." (page 229).

LD 16 Configure the SIP route.

Prompt	Response	Description
REQ	NEW	Add a new route.
TYPE	RDB	Route Data Block

Prompt	Response	Description
CUST	xx	Customer number as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system
	0-127	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T
DES	x...x	Designator
		The designator field for the trunk groups. This designator can be 0-16 alphanumeric characters.
TKTP		Trunk Type
	TIE	TIE trunk
VTRK		Virtual Trunk route, where:
	YES	YES = This route is for Virtual Trunk
		NO = This route is not for Virtual Trunk (default)
ZONE	0-255	Zone for codec selection and bandwidth management
PCID	SIP	Protocol ID for the SIP route.
		Defines the route as a SIP route.
CRID	(NO) YES	CDR record (for SIP) to include correlation ID.
		YES = When enabled, the fourth line is included in the new CDR record. NO = The fourth line in not included in the CDR record (default).
		See *Call Detail Recording: Description and Formats* (NN43001-550) for more information.
		Note: This prompt appears only for a SIP Virtual Trunk (that is, if VTRK = YES and PCID = SIP) and CDR is turned on for this route.
NODE	xxxx	Node ID
		Where the Node ID matches the node of the Signaling Server. The Node ID can have a maximum of four numeric characters.
ISDN	YES	Integrated Services Digital Network option
- MODE	ISLD	Mode of operation
- DCH	0-159	D-channel number
- IFC	SL1	Interface type for route (IFC responses are listed in *Software Input/Output: Administration* [NN43001-611])
- SRVC	a...a	Service type for AT&T ESS connections (SRVC responses are listed in *Software Input/Output: Administration* [NN43001-611])

Prompt	Response	Description
- - PNI	(0)-32700	Private Network Identifier
- NCNA	(YES)	Network Calling Name Allowed
- NCRD	YES	Network Call Redirection
- INAC	(NO) YES	Inserts the ESN access code to an incoming private network call. INAC enables an ESN access code to be automatically added to an incoming ESN call from a private network. If INAC = YES, then digit insertion (INST) for NARS or BARS calls is bypassed and Access Code 1 (AC1) is used for all call types. However, calls can be specifically defined to use Access Code 2 (AC2) in LD 15 at the AC2 prompt. INAC is prompted when the route type is either a TIE trunk or an IDA trunk with DPNSS1 signaling.
ICOG		Incoming and Outgoing trunk.
	IAO	Incoming and Outgoing
ACOD	x...x	Access Code for the trunk route.

LD 16 Configure the H.323 route.

Prompt	Response	Description
REQ	NEW	Add a new route.
TYPE	RDB	Route Data Block
CUST	xx	Customer number as defined in LD 15.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system
	0-127	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T
DES	x...x	Designator The designator field for the trunk groups. This designator can be 0-16 alphanumeric characters.
TKTP		Trunk Type
	TIE	TIE trunk
VTRK		Virtual Trunk route, where:
	YES	YES = This route is for Virtual Trunk NO = This route is not for Virtual Trunk (default)
ZONE	0-255	Zone for codec selection and bandwidth management
NODE	xxxx	Node ID Where the Node ID matches the node of the Signaling Server. The Node ID can have a maximum of four numeric characters.

Prompt	Response	Description
PCID	H323	Protocol ID for the H.323 route.
ISDN	YES	Integrated Services Digital Network option
- MODE	ISLD	Mode of operation
- DCH	0-159	D-channel number
- IFC	SL1	Interface type for route (IFC responses are listed in *Software Input/Output: Administration* [NN43001-611])
- SRVC	a...a	Service type for AT&T ESS connections (SRVC responses are listed in *Software Input/Output: Administration* [NN43001-611])
- - PNI	(0)-32700	Private Network Identifier
- NCNA	(YES)	Network Calling Name Allowed
- NCRD	YES	Network Call Redirection
- INAC	(NO) YES	Inserts the ESN access code to an incoming private network call. INAC enables an ESN access code to be automatically added to an incoming ESN call from a private network. If INAC = YES, then digit insertion (INST) for NARS or BARS calls is bypassed and Access Code 1 (AC1) is used for all call types. However, calls can be specifically defined to use Access Code 2 (AC2) in LD 15 at the AC2 prompt. INAC is prompted when the route type is either a TIE trunk or an IDA trunk with DPNSS1 signaling.

LD 97 Configure the superloop for the Virtual Trunks.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	SUPL	Superloop
SUPL	 0-159 0-255	Superloop number 0-159: Superloop number in multiples of 4 0-255: Systems with Fiber Network Fabric

LD 14 Configure Virtual Trunks.

Prompt	Response	Description
REQ	NEW NEW x	Create a trunk Create x trunks, where x = 1-255 (to create that number of consecutive trunks)
TYPE	IPTI	IP TIE trunk data block
TN		Terminal Number

Prompt	Response	Description
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit
	c u	Format for Small System, Media Gateway 1000B, and Media Gateway 1000T,where c = card and u = unit
DES	a....a	Virtual Trunk descriptor
		Designator field for trunk groups where a...a = 0-16 alphanumeric characters (DES is an optional entry)
XTRK		Extended Trunk
	VTRK	Virtual Trunk type
		Note: If you entered a virtual TN at the TN prompt, then the XTRK prompt only accepts the VTRK option.
CUST	xx	Customer number as defined in LD 15.
...		
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system
	0-127 1-4000	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T
CHID	1-4300	Channel ID for this trunk, dependent on the ISLM parameter (LD 17)
STRI		Start arrangement Incoming
	IMM	Immediate
STRO		Start arrangement Outgoing
	IMM	Immediate
SUPN	YES	Answer and disconnect Supervision required
		SUPN must equal YES for a COT with Virtual Network Service
...		
TKID	nnnnnnn	Trunk Identifier

LD 86 Configure dialing plan, networking, and ESN data.

Prompt	Response	Description
REQ	NEW	Create new data block
FEAT	ESN	Electronic Switched Network

Prompt	Response	Description
MXLC	0-999 0-16000	Maximum number of Location Codes (NARS only) Maximum number of Location Codes (NARS only) (with the ESN Location Code Expansion feature and the FNP feature enabled). Refer to ESN Location Code Expansion feature in *ISDN Primary Rate Interface: Features* NN43001-569-B1.
...		
CDP	YES	Coordinated Dialing Plan feature for this customer
- MXSC	x	Maximum number of Steering Codes Where x = • 0-8000 = Maximum number of Steering Codes for Small Systems • 0-10000 = Maximum number of Steering Codes in North America • 0-32000 = Maximum number of Steering Codes outside North America
- NCDP	x	Number of digits to be included as part of the CDP DN (DSC + DN or LSC + DN) where x = 3-7.
AC1	x	One- or two-digit NARS/BARS Access Code 1
AC2	x	One- or two-digit NARS Access Code 2
DLTN	(YES)	NARS/BARS Dial Tone after dialing AC1 or AC2 access codes
ERWT	(YES)	Expensive Route Warning Tone
...		
TGAR	(NO)	Check for Trunk Group Access Restriction.

LD 87 Configure network access.

Prompt	Response	Description
REQ	NEW	Add new data.
FEAT	NCTL	Network Control Block
SOHQ	(NO)	Off-Hook Queuing option
SCBQ	(NO)	Call-Back Queuing option

Prompt	Response	Description
NCOS	(0)	Network Class of Service group number
TOHQ	(0)	TCOS OHQ eligibility

LD 86 Configure the Digit Manipulation Index.

Prompt	Response	Description
REQ	NEW	Create new data.
CUST	xx	Customer number as defined in LD 15.
FEAT	DGT	Digit manipulation data block
DMI	xxxx	Digit Manipulation Index numbers Digit Manipulation Index with Flexible Numbering Plan (FNP) package 160 DMI is only prompted when the Directory Number Expansion (DNXP) package 150 is equipped and SDRR = LDID.
DEL	xx	Delete Number of leading digits to be deleted
INST	<cr>	Insert Up to 31 leading digits can be inserted
CTYP	<cr>	Call Type to be used by the manipulated digits. This call type must be recognized by the far-end switch.
...		

Nortel recommends that all routes in a Route List Block (RLI) be configured as either overlap or en bloc. That is, an en bloc route should not have alternate routes that are configured as overlap, and vice versa. Erratic behavior can occur when overlap and en bloc routes are configured as alternate routes. Normal behavior occurs on alternate routes as long as the alternate route has the same overlap capabilities as the main route.

LD 86 Configure the Route List Block for the Virtual Trunk route.

Prompt	Response	Description
REQ	NEW	Create new data block
FEAT	RLB	Route list block
...		
RLI	 0-127 0-255 0-999	Route List Index to be accessed CDP and BARS NARS FNP

Prompt	Response	Description
ENTR	xxx	Entry number for NARS/BARS Route list Where xxx = • 0-63 Entry number for NARS/BARS Route List • 0-6 Route list entry number for CDP • X Precede with x to remove
LTER	(NO)	Local Termination entry
ROUT		Route number
	0-511	Range for Large System and CS 1000E system
	0-127	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T
DMI		Digit Manipulation Index
	0	No digit manipulation required
	1-31	CDP
	0-255	NARS and BARS
	0-999	FNP
...		

LD 87 Configure the CDP steering codes.

Prompt	Response	Description
REQ	NEW	Create new data block
FEAT	CDP	Coordinated Dialing Plan
TYPE		Type of steering code
	DSC	Distant Steering Code
DSC	x..x	Distant Steering Code Up to 4 digits; up to 7 digits with Directory Number Expansion (DNXP) package 150.
- FLEN	(0)	Flexible Length number of digits
- DSP	(LSC)	Display (Local Steering Code)
- RRPA	(NO)	Remote Radio Paging Access
- RLI		Route List Index to be accessed for Distant Steering Code. Cannot use non-zero entries or DMI.
	0-31	CDP
	0-127	BARS
	0-255	NARS
	0-999	Flexible Numbering Plan (FNP)
- CCBA	(NO)	Collect Call Blocking (CCB) Denied

Prompt	Response	Description
- NPA	<cr>	North American Numbering Plan Routing code: maximum 7-digit National code enabled
- NXX	<cr>	North American Numbering Plan Routing code: maximum 7-digit subscriber code allowed

LD 90 Configure call types and Location Codes.

Prompt	Response	Description
REQ	NEW CHG	Create new data block Change existing data block
CUST	xx	Customer number as defined in LD 15.
FEAT	NET	Network Translator (Network translation tables)
TRAN	 AC1 AC2	Translator Access Code 1 (NARS/BARS) Access Code 2 (NARS)
TYPE	LOC	Location Code
LOC	x...x	Location Code
- FLEN	(0)-10	Flexible Length
		Enter the maximum number of digits expected. When this number of digits is dialed, dialing is considered to be complete and end-of-dial processing begins.
		Default is zero (0) digits.
- RLI	0-999	Route List Index
		Enter Route List Index for this LOC.
...		

VNR enhancement

To configure the VNR enhancement, configure AC2, PFX1, VNR, RLI, CDPL, UDPL, CNTC, CATC, and INTC in LD 15.

LD 15 Configure the VNR enhancement.

Prompt	Response	Description
REQ:	NEW	Add new data block to the system.
TYPE:	NET	ISDN and ESN networking options
CUST		Customer number
	0-99	Range for Large System and CS 1000E system
	0-31	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T

Prompt	Response	Description
OPT		Options
	RTD	Coordinated Dialing Plan routing feature Denied
AC2	SPN LOC	Special Number; Location Code
FNP	(YES)	Enable Flexible Numbering Plan for customer.
ISDN	YES	Integrated Services Digital Network allowed for customer. ***Note:*** Prompted when ISDN signaling package 145 is equipped and either the Integrated Service Digital Network BRI Trunk Access (BRIT) package 233 is equipped or at least one PRA link is configured.
- VPNI	1-16283	Virtual Private Network Identifier
- CLID	YES	Allow Calling Line Identification option Calling Line Identification does not require ISDN.
- - ENTRY	xx	CLID entry to be configured. CLID entries must be between 0 and the value entered at the SIZE prompt - 1. Precede entry or entries with X to delete. ENTRY is repeated until a <cr> is entered.
- - - HLOC	100-9999999	Home Location Code (ESN) as defined in LD 90 1 to 7 digits with extended code. Prompted when ISDN=YES, or with Digital Private Network Signaling System 1 (DPNSS) package 123.
- - - LSC	0 .. x..x	Local Steering Code 1 to 7 digits. LSCs are required if the CDP DNs are longer than the local PDNs. The CLID sent for a CDP call is composed of the LSC defined in LD 15 plus the PDN of the calling set. Various ISDN network features depend on the CLID as the return address for sending feature control messages. Multiple LSCs may be defined in LD 87 for CDP but only one LSC can be defined here for the CLID. The LSC prompt appears only if the user has a five or six digit dialing plan, or if the DPNSS software package is equipped. LSC is prompted here if ISDN = NO, otherwise LSC is a sub-prompt of ISDN.
- PFX1	xxxx	Prefix 1. Prefix or area code for International PRA. First element of Calling Party Number. PFX1 + PFX2 + DN cannot exceed 8 numbers for AXE-10. Prompted with International Primary Rate Access (IPRA) package 202.

Prompt	Response	Description
- PRX2	xxxx	Prefix 2. Central Office Prefix for International PRA. Second element of Calling Part Number. PFX1 + PFX2 + DN cannot exceed 8 numbers for AXE-10. Prompted with International Primary Rate Access (IPRA) package 202.
- RCNT	0-(5)	Redirection Count for ISDN calls Maximum number of inter-node hops allowed in a network redirection call, only enforced when ISDN = YES. This field must be set to greater than 0 for a network redirection to take place.
- PSTN	(NO)	Public Service Telephone Networks Limit the number of PSTNs allowed in a network connection to one PSTN. The default (NO) puts no limit on the number of PSTN connections.
- - TNDM	0-(15)-31	Tandem Threshold/Loop Avoidance Limit This is the value permitted in a network connection. If the value entered is greater than 25, then 25 will be used for DPNSS calls. Prompted when Integrated Services Digital Network (ISDN) package 245 and ISDN Supplementary Features (ISDN INTL SUP) package 161, or Digital Private Signaling System Network Services (DNWK) package 231 is equipped.
- - PCMC	0-(15)-31	Pulse Code Modulation Conversions permitted in a network connection, µ-Law to A- Law or A- Law to µ-Law, in a network connection
- SATD	0-(1)-5	Satellite Delays. Number of satellite delays allowed in a network connection
OCLI	NO	NO manipulation is done on outgoing CLID for calls forwarded to EuroISDN link.
TIDM	(NO)	Trunk Identity Meaningful
DASC	xxxx	Display Access Code Enter the access code which is to be placed on displays before Originating Line Identities (OLI) and Terminating Line Identities (TLI) are received from the ISDN. The default is no code, when creating a new data block. Prompted with Multi Language Wake Up (MLWU) package 206 and Integrated Digital Access (IDA) package 122.

Prompt	Response	Description
ROPT	(NRO)	No Route Optimization This option may be used to suppress Route Optimization on switches which already have high traffic.
DITI	(NO)	DID to TIE connections allowed
TRNX	(NO)	Prevent transfer on ringing of supervised external trunks across a private network
EXTT	(NO)	Prevent connection of supervised external trunks via either call transfer or conference
FTOP	(FRES)	Flexible Trunk to Trunk Options. Flexible Trunk to Trunk Connections Restricted. FTT feature is inactive.
APAD	x y (0) (0)	Alternative Pad. Where: x = trunk pad selection and y = conference pad selection Valid inputs for x are: (0) = default North America 1 = Australia 2 = New Zealand 3 = Italy 4 = China EPE or EPE/IPE systems 5 = China pure IPE system 6-7 = future usage currently set to default Valid inputs for y are: (0) = default North America 1 = Alternative Conference pads selected The default = 0 when REQ = NEW. The default is the existing value when REQ = CHG.Alternative Conference pads are only provided on specific Conference cards.
DMWM	(NO)	Enable the output of DPNSSI Message Waiting Indication Non Specified Information error messages
MWNS	(NO)	Message Waiting Indication DPNSSI Non Specified Information string to recognize.
VNR	(YES)	Vacant Number Routing
- RLI	0-999	Route List Index as defined in LD 86
- CDPL	1-(10)	Flexible length of Vacant Number Routing (VNR) Coordinated Dialing Plan (CDP)

Prompt	Response	Description
- UDPL	1-(19)	Uniform Plan Public Flexible length of Vacant Number Routing (VNR) Uniform Dialing Plan digits (UDP). Enter the maximum number of UDP digits expected by VNR.
NIT	2-(8)	Network Alternate Route Selection (NARS) Interdigit Timer
NAS_ATCL	(YES)	Network Attendant Service Attendant Control allowed
NAS_ACTV	NO	Network Attendant Service routing Activated
FOPT	0-(6)-30	Flexible Orbiting Prevention Timer The number of seconds in two second intervals that CFW should be suspended on a set that has just forwarded a call off-node. Odd entries are rounded up to the next valid entry. A response of 0 disables FOPT.
CNDN	0 .. x..x	Customer Calling Number Identification DN on outgoing Multifrequency Compelled Signaling (MFC) calls
- CNIP	(YES)	Calling Number Identification Presentation Send Customer Calling Number Identification (CNDN) + Trunk ID (TKID) if Calling Line ID (CLID) = NO in LD 17
CNAT	0 .. x..x	CNI Attendant DN on outgoing Multifrequency Compelled Signaling (MFC) calls.
CNTC	x	Country Code (see Note 1)
NATC	x	National Access Code (see Note 1)
INTC	xxx	International Access Code (see Note 1)

LD 21 prints which dialing plan is used with AC1. This helps identify which dialing plans use AC1 and which other dialing plans use AC2.

LD 21 - Print the dialing plan.

Prompt	Response	Description
REQ	PRT	Print data block for the TYPE specified
TYPE	NET	ISDN and ESN networking options
CUST	xx	Customer number as defined in LD 15.

Configuring the Gateways

Both H.323 Gateways and SIP Trunk Gateways (that is, the Virtual Trunk applications) are supported.

The four possible configurations are:

- no Gateways (no Virtual Trunk)
- H.323 Gateway only (H.323 Virtual Trunk only)
- SIP Trunk Gateway only (SIP Virtual Trunk only)
- Both H.323 and SIP Trunk Gateways (both H.323 and SIP Virtual Trunks)

Enabling and configuring the H.323 Gateway

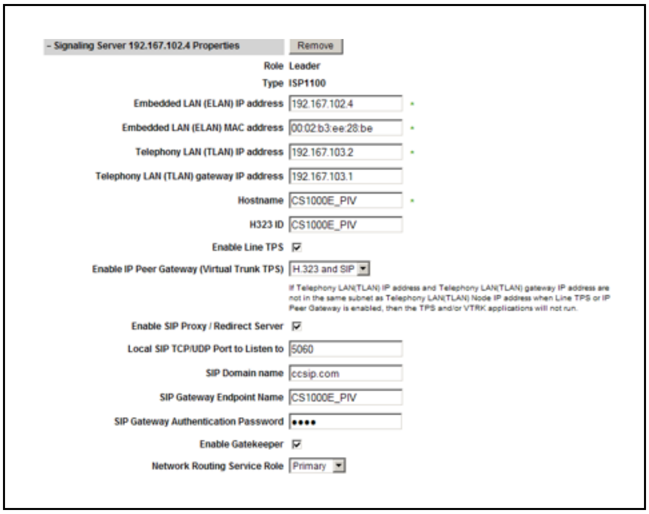
The H.323 Gateway runs only on the Signaling Server. However, configuration of the H.323 Gateway requires configuration on both the Call Server and the Signaling Server. You must use Element Manager to configure the H.323 Gateway on the Signaling Server.

- For Call Server configuration, follow "Feature Implementation of IP Peer Networking" (page 224). In LD 16, configure the route as an H.323 route (see "LD 16 – Configure the H.323 route." (page 229)).
- For Signaling Server configuration, perform the following procedures using Element Manager:
  - Procedure 19 "Enabling the H.323 Gateway (H.323 Virtual Trunk application)" (page 240)
  - Procedure 20 "Configuring the H.323 Gateway settings" (page 241)

Procedure 19
Enabling the H.323 Gateway (H.323 Virtual Trunk application)

Step	Action
1	Log in to Element Manager.
2	Select **System > IP Network > Nodes: Servers, Media Cards** from the EM Navigator. The **Node Configuration** web page opens, as shown in Figure 110 "Node Configuration web page" (page 206).
3	Click **Edit**. The **Edit** web page opens, as shown in Figure 111 "Edit web page" (page 207).
4	Click **Signaling Servers** to expand the section. A list of Signaling Servers opens.
5	Select the appropriate **Signaling Server xxx.xxx.xxx.xxx Properties**. The properties for that Signaling Server display, as shown Figure 135 "Signaling Server xxx.xxx.xxx.xxx properties" (page 241).

Figure 135
Signaling Server xxx.xxx.xxx.xxx properties



- Signaling Server 192.167.102.4 Properties [Remove]

Role	Leader
Type	ISP1100
Embedded LAN (ELAN) IP address	192.167.102.4 *
Embedded LAN (ELAN) MAC address	00:02:b3:ee:28:be *
Telephony LAN (TLAN) IP address	192.167.103.2 *
Telephony LAN (TLAN) gateway IP address	192.167.103.1
Hostname	CS1000E_PIV *
H323 ID	CS1000E_PIV
Enable Line TPS	☑
Enable IP Peer Gateway (Virtual Trunk TPS)	H.323 and SIP
	If Telephony LAN(TLAN) IP address and Telephony LAN(TLAN) gateway IP address are not in the same subnet as Telephony LAN(TLAN) Node IP address when Line TPS or IP Peer Gateway is enabled, then the TPS and/or VTRK applications will not run.
Enable SIP Proxy / Redirect Server	☑
Local SIP TCP/UDP Port to Listen to	5060
SIP Domain name	ccsip.com
SIP Gateway Endpoint Name	CS1000E_PIV
SIP Gateway Authentication Password	••••
Enable Gatekeeper	☑
Network Routing Service Role	Primary

6 Select an **H.323 option** from the **Enable IP Peer Gateway (Virtual Trunk TPS)** drop-down list.

This field is used to enable H.323 Gateway.

> ***Note:*** The four supported modes are: None, H.323 only, SIP only, and H.323 and SIP.

7 Verify the **H323 ID**. Each H.323 Gatekeeper is configured with an H.323 Gatekeeper alias name, which is an H323-ID. Enter any text string to describe the H.323 Virtual Trunk source in the **H323 ID** text box.

8 Click **Save and Transfer**.

—End—

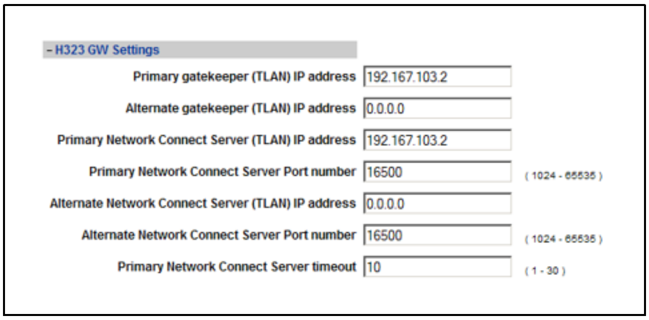
Procedure 20
Configuring the H.323 Gateway settings

Step	Action
1	Log in to Element Manager.
2	Select **System > IP Network > Nodes: Servers, Media Cards** from the EM Navigator. The **Node Configuration** web page opens, as shown in Figure 110 "Node Configuration web page" (page 206).
3	Click **Edit**.

The **Edit** web page opens, as shown in Figure 111 "Edit web page" (page 207).

4 Select **H323 GW Settings** to expand the section, as shown in Figure 136 "H323 GW Settings" (page 242).

Figure 136
H323 GW Settings



– H323 GW Settings

Primary gatekeeper (TLAN) IP address: 192.167.103.2

Alternate gatekeeper (TLAN) IP address: 0.0.0.0

Primary Network Connect Server (TLAN) IP address: 192.167.103.2

Primary Network Connect Server Port number: 16500 (1024 - 65535)

Alternate Network Connect Server (TLAN) IP address: 0.0.0.0

Alternate Network Connect Server Port number: 16500 (1024 - 65535)

Primary Network Connect Server timeout: 10 (1 - 30)

5 Configure the following fields:

a. **Primary gatekeeper IP address:** Enter the TLAN network interface IP address (not the Node IP address) of the Leader Signaling Server running the H.323 Gatekeeper.

b. **Alternate gatekeeper IP address:** Enter the IP address if an Alternate Gatekeeper exists.

c. **Primary Network Connect Server IP address:** Enter or verify that the NCS IP address matches the Primary gatekeeper IP address (NRS). The NCS is used for IP Line Virtual Office, Branch Office (including the SRG), and Geographic Redundancy features. The NCS allows the Line TPS (LTPS) to query the NRS using the UNIStim protocol.

d. **Primary Network Connect Server Port number:** Enter a port number for the Primary NCS. The port number must be numeric and up to 5 numbers in length. The range is 1024 to 65535. The default value is 16500.

e. **Alternate Network Connect Server IP address:** Enter the IP address of the alternate NCS IP address.

f. **Alternate Network Connect Server Port number:** Enter a port number for the Alternate NCS. The port number must be numeric and up to 5 numbers in length. The range is 1024 to 65535. The default value is 16500.

g. **Primary Network Connect Server timeout:** Enter a timeout value for the Primary NCS. The range is 1 to 30 seconds. The default value is 10 seconds.

6 Click **Save and Transfer**.

—End—

Enabling and configuring the SIP Trunk Gateway

The SIP Trunk Gateway runs only on the Signaling Server. Configuration of the SIP Trunk Gateway requires configuration on both the Call Server and the Signaling Server. You must use Element Manager to configure the SIP Trunk Gateway on the Signaling Server.

- For Call Server configuration, follow "Feature Implementation of IP Peer Networking" (page 224). In LD 16, configure the route as a SIP route.
- For Signaling Server configuration, perform the following procedures using Element Manager:
  - — Procedure 21 "Enabling the SIP Trunk Gateway (SIP Virtual Trunk application)" (page 243)
  - — Procedure 22 "Configuring the SIP Trunk Gateway settings" (page 245)
  - — Procedure 23 "Configuring the SIP URI to NPI/TON mapping" (page 247)

Procedure 21

Enabling the SIP Trunk Gateway (SIP Virtual Trunk application)

Step Action

1 Log in to Element Manager.

2 Select **System > IP Network > Nodes: Servers, Media Cards** from the EM Navigator.

The **Node Configuration** web page opens, as shown in Figure 110 "Node Configuration web page" (page 206).

3 Click **Edit**.

The **Edit** web page opens, as shown in Figure 111 "Edit web page" (page 207).

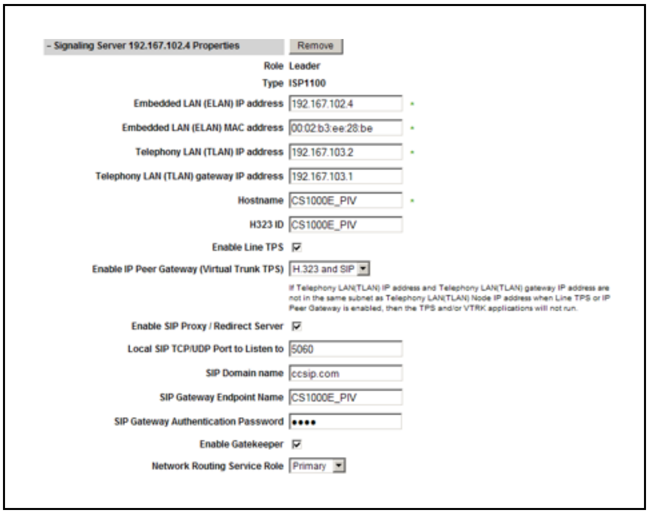
4 Select **Signaling Servers** to expand the section.

A list of Signaling Servers opens.

5 Select the appropriate **Signaling Server xxx.xxx.xxx.xxx Properties**.

The properties for that Signaling Server display, as shown in Figure 137 "Signaling Server xxx.xxx.xxx.xxx properties" (page 244).

Figure 137
Signaling Server xxx.xxx.xxx.xxx properties



- Signaling Server 192.167.102.4 Properties [Remove]

Role	Leader
Type	ISP1100
Embedded LAN (ELAN) IP address	192.167.102.4 *
Embedded LAN (ELAN) MAC address	00:02:b3:ee:28:be *
Telephony LAN (TLAN) IP address	192.167.103.2 *
Telephony LAN (TLAN) gateway IP address	192.167.103.1
Hostname	CS1000E_PIV *
H323 ID	CS1000E_PIV
Enable Line TPS	☑
Enable IP Peer Gateway (Virtual Trunk TPS)	H.323 and SIP
	If Telephony LAN(TLAN) IP address and Telephony LAN(TLAN) gateway IP address are not in the same subnet as Telephony LAN(TLAN) Node IP address when Line TPS or IP Peer Gateway is enabled, then the TPS and/or VTRK applications will not run.
Enable SIP Proxy / Redirect Server	☑
Local SIP TCP/UDP Port to Listen to	5060
SIP Domain name	ccsip.com
SIP Gateway Endpoint Name	CS1000E_PIV
SIP Gateway Authentication Password	••••
Enable Gatekeeper	☑
Network Routing Service Role	Primary

6 Select a **SIP option** from the **Enable IP Peer Gateway (Virtual Trunk TPS)** drop-down list.

This field is used to enable SIP Trunk Gateway and Services.

> ***Note:*** The four supported modes are: None, H.323 only, SIP only, and H.323 and SIP.

7 Select the **SIP Transport Protocol**. This is the transport protocol used for SIP message exchange between the Gateway and Redirect/Proxy Server. The two options are TCP and UDP. TCP is the default option.

> ***Note:*** Nortel recommends that you use the default option (TCP) for SIP traffic.

8 Verify the **Local SIP Port**. This is the port to which the gateway listens. The default is 5060.

9 Enter the **SIP Domain Name**. This string identifies the SIP Service Domain. The SIP Domain Name configured in the Signaling Server properties must match the Service Domain name configured in the NRS (see *Adding a Service Domain*). This string is used in building all SIP messages and appears in the phone context. The string must be less than 128 characters in length. The valid characters are a-z, 0-9, period (.), hyphen (-), comma (,), and underscore (_). This field must be specified if the SIP Trunk Gateway application is enabled.

10 If authentication is turned on in the NRS (SIP Redirect Server) or on the MCS 5100 Proxy Server, then the **SIP Gateway Endpoint Name** and **SIP Gateway Authentication Password** must be entered and must match the Gateway Endpoint name and Gateway

Endpoint authentication password used by the SIP Redirect Server (see *Adding a Gateway Endpoint*). The name and authentication password are used in authenticating the Gateway Endpoint with the SIP Redirect Server.

a. **SIP Gateway Endpoint Name:** Enter the endpoint name. This is the username that is used when authenticating this gateway with the NRS (SIP Redirect Server) or the MCS 5100 Proxy Server. This field must be specified if authentication is enabled for the Gateway Endpoint in the NRS or Proxy Server.

b. **SIP Gateway Authentication Password:** Enter the password. This is the password that is used when authenticating this gateway with the NRS (SIP Redirect Server) or the MCS 5100 Proxy Server. This field must be specified if authentication is enabled for the Gateway Endpoint in the NRS or Proxy Server.

11 Click **Save and Transfer**.

—End—

Procedure 22
Configuring the SIP Trunk Gateway settings

Step Action

1 Log in to Element Manager.

2 Select **System IP Network > Nodes: Servers, Media Cards** from the EM Navigator.

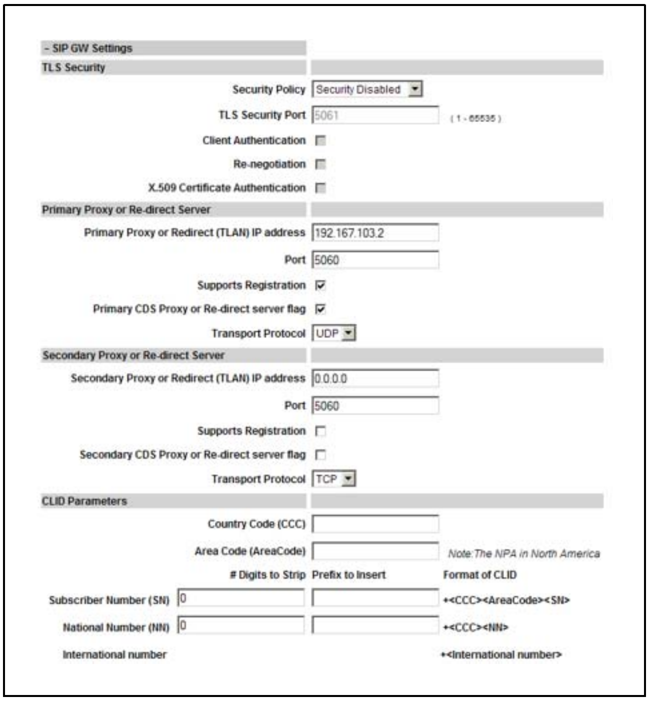
The **Node Configuration** web page opens, as shown in Figure 110 "Node Configuration web page" (page 206).

3 Click **Edit**.

The **Edit** web page opens, as shown in Figure 111 "Edit web page" (page 207).

4 Select **SIP GW Settings** to expand the section (see Figure 138 "SIP GW Settings" (page 246)).

Figure 138
SIP GW Settings



- SIP GW Settings

TLS Security

Security Policy: Security Disabled ▾

TLS Security Port: 5061 (1 - 65535)

Client Authentication □

Re-negotiation □

X.509 Certificate Authentication □

Primary Proxy or Re-direct Server

Primary Proxy or Redirect (TLAN) IP address: 192.167.103.2

Port: 5060

Supports Registration ☑

Primary CDS Proxy or Re-direct server flag ☑

Transport Protocol: UDP ▾

Secondary Proxy or Re-direct Server

Secondary Proxy or Redirect (TLAN) IP address: 0.0.0.0

Port: 5060

Supports Registration □

Secondary CDS Proxy or Re-direct server flag □

Transport Protocol: TCP ▾

CLID Parameters

Country Code (CCC):

Area Code (AreaCode): *Note:The NPA in North America*

	# Digits to Strip	Prefix to Insert	Format of CLID
Subscriber Number (SN)	0		+<CCC><AreaCode><SN>
National Number (NN)	0		+<CCC><NN>
International number			+<International number>

5 Complete the following for the Primary server:

a. **TLS Security:** Select the **Security Policy** from the drop-down list.

b. **TLS Security Port:** Enter a value for the port in the text box.

c. **Client Authentication:**

d. **Re-negotiation:**

e. **X.500 Certificate Authentication:**

f. **Primary Proxy / Re-direct IP address:** Enter the TLAN network interface IP address of the Primary SIP Redirect Server or the MCS 5100 Proxy Server.

g. **Primary Proxy / Re-direct IP Port:** Leave the default port value as 5060 for the Primary SIP Redirect Server or the MCS 5100 Proxy Server.

h. **Primary Proxy Supports Registration:** This check box tells the SIP Trunk Gateway whether the primary NRS (SIP Redirect Server) supports registration. If the check box is selected, then the SIP Trunk Gateway must register with the primary NRS. If the check box is not selected, then the SIP Trunk Gateway will not register with the primary NRS.

i. The **Primary CDS Proxy or Re-direct server flag** is not used in this release.

j. **Secondary Proxy / Re-direct IP address:** Enter the TLAN network interface IP address of the Secondary SIP Redirect Server or the MCS 5100 Proxy Server (if configured).

k. **Secondary Proxy / Re-direct IP Port:** Leave the default port value as 5060 for the Secondary SIP Redirect Server or the MCS 5100 Proxy Server (if configured).

l. **Secondary Proxy Supports Registration:** This check box tells the SIP Trunk Gateway whether the secondary NRS (SIP Redirect Server) supports registration. If the check box is selected, then the SIP Trunk Gateway must register with the secondary NRS. If the check box is not selected, then the SIP Trunk Gateway will not register with the secondary NRS.

m. The **Secondary CDS Proxy or Re-direct server flag** is not used in this release.

6 Click **Save and Transfer**.

—End—

Configuring the SIP URI to NPI/TON mapping using Element Manager

The SIP URI to NPI/TON mapping is used as a translation of a signaling request between the SIP Trunk Gateway and the NRS.

The SIP Trunk Gateway sends a request to the NRS to find the SIP address resolution. To configure the SIP Trunk Gateway to communicate with the NRS (SIP Redirect Server), the SIP URI to NPI/TON mapping must be done.

Once the NRS server is properly configured properly and the NRS numbering plan database had been provisioned (see *Network Routing Service Installation and Commissioning (NN43001-564)*), you must build the SIP URI to NPI/TON mapping using Element Manager.

Procedure 23 "Configuring the SIP URI to NPI/TON mapping" (page 247) provides the steps to create this SIP URI to NPI/TON mapping using an NRS example and an example for the MCS 5100.

Procedure 23
Configuring the SIP URI to NPI/TON mapping

Step	Action
1	Log in to Element Manager.

2 Select **System IP Network > Nodes: Servers, Media Cards** from the EM Navigator.

The **Node Configuration** web page opens, as shown in Figure 110 "Node Configuration web page" (page 206).

3 Click **Edit**.

The **Edit** web page opens, as shown in Figure 111 "Edit web page" (page 207).

4 Select **SIP URI Map** to expand the section.

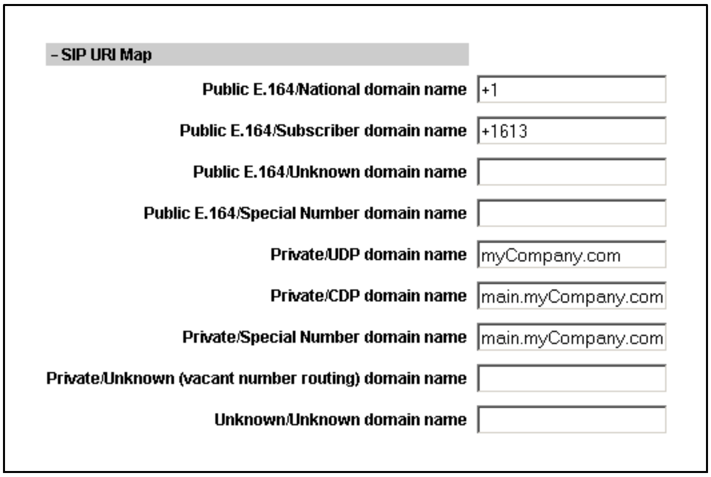
> ***Note:*** The fields require a character string that is less than 128 characters in length. The valid characters include: a-z, 0-9, ., -, _, and +. These fields must be completed if the SIP Trunk Gateway application is enabled.

The values in this SIP URI Map section are based on the example provided in the *Network Routing Service overview* chapter of *Network Routing Service Installation and Commissioning (NN43001-564)* , specifically the examples provided in *Numbering plan mapping*.

To complete the NRS example, refer to Figure 139 "SIP URI Map for the NRS example" (page 248) and go to Step 5.

To complete the MCS 5100 example, refer to Figure 140 "SIP URI Map for the MCS 5100 example" (page 249) and go to Step 6.

Figure 139
SIP URI Map for the NRS example



- SIP URI Map

Public E.164/National domain name	+1
Public E.164/Subscriber domain name	+1613
Public E.164/Unknown domain name	
Public E.164/Special Number domain name	
Private/UDP domain name	myCompany.com
Private/CDP domain name	main.myCompany.com
Private/Special Number domain name	main.myCompany.com
Private/Unknown (vacant number routing) domain name	
Unknown/Unknown domain name	

5 Fill in the following fields for the NRS example (see "Configuring call routing" (page 202)):

a. Type **+1** in the **Public E.164/National domain name** text box.

b. Type **+1613** in the **Public E.164/Subscriber domain name** text box.

c. Leave the **Public E.164/Unknown domain name** text box blank.

d. Leave the **Public E.164/Special Number domain name** text box blank

e. Type **myCompany.com** in the **Private/UDP domain name** text box.

f. Type **myCdpDomain.myCompany.com** in the **Private/CDP domain name** text box.

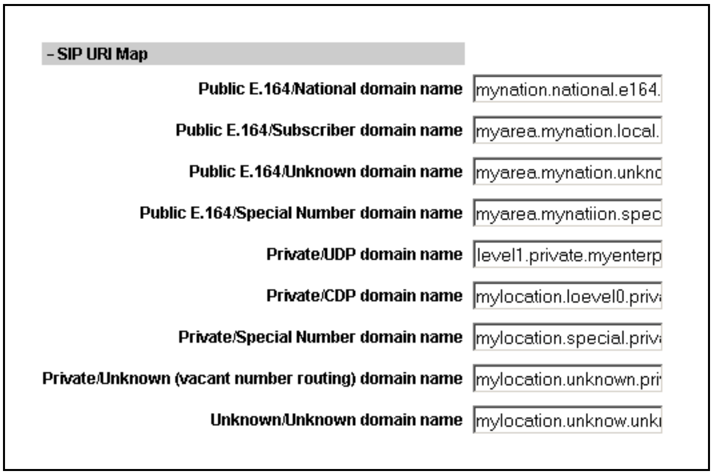
g. Type **special.myCdpDomain.myCompany.com** in the **Private/Special Number domain name** text box.

h. Leave the **Private/Unknown (vacant number routing) domain name** text box blank.

i. Leave the **Unknown/Unknown domain name** text box blank.

j. Click **Save and Transfer**.

Figure 140
SIP URI Map for the MCS 5100 example



– SIP URI Map

Public E.164/National domain name	mynation.national.e164.
Public E.164/Subscriber domain name	myarea.mynation.local.
Public E.164/Unknown domain name	myarea.mynation.unknc
Public E.164/Special Number domain name	myarea.mynatiion.spec
Private/UDP domain name	level1.private.myenterp
Private/CDP domain name	mylocation.loevel0.priv
Private/Special Number domain name	mylocation.special.priv
Private/Unknown (vacant number routing) domain name	mylocation.unknown.pri
Unknown/Unknown domain name	mylocation.unknow.unk

6 Fill in the following fields for the MCS 5100 example (see Figure 140 "SIP URI Map for the MCS 5100 example" (page 249)):

a. Type **mynation.national.e164.myrootdomain** in the Public E.164/National domain name text box.

b. Type **myarea.mynation.local.e164.myrootdomain** in the Public E.164/Subscriber domain name text box.

c. Type **myarea.mynation.unknown.e164.myrootdomain** in the Public E.164/Unknown domain name text box.

d. Type **myarea.mynation.special.e164.myrootdomain** in the Public E.164/Special Number domain name text box.

e. Type **level1.private.myenterprise** in the Private/UDP domain name text box.

f. Type **mylocation.level0.private.myenterprise** in the Private/CDP domain name text box.

g. Type **mylocation.special.private.myenterprise** in the Private/Special Number domain name text box.

h. Type **mylocation.unknown.private.myenterprise** in the Private/Unknown (vacant number routing) domain name text box.

i. Type **mylocation.unknown.unknown.myrootdomain** in the Unknown/Unknown domain name text box.

j. Click **Save and Transfer**.

—End—

Restarting the Signaling Server

Some fields in Element Manager can be changed at run-time: SIP domain name, CDS proxy (yes or no), Gateway username and password, dialing plans, and all "SIP Service" related fields except ACD DN. The rest of the fields require a restart of the Signaling Server.

Warm restart

To warm restart the Signaling Server, following the steps in Procedure 24 "Warm restarting the Signaling Server" (page 250).

Procedure 24
Warm restarting the Signaling Server

Step	Action
1	Select **System > IP Network > Maintenance and Reports** from the EM Navigator.
2	Select the node containing the Signaling Server to be restarted.
3	Click **Reset** for the Signaling Server.

—End—

Cold restart

Press the **RST** button on the front panel to cold restart the Signaling Server.

Overlap signaling

Contents

This section contains information on the following topics:

Overview

Overlap signaling over IP is supported using the H.323 protocol.

> ***Note:*** Overlap signaling is not supported using the Session Initiation Protocol (SIP).

Both overlap signaling and en bloc signaling is supported. The difference between overlap and en bloc signaling is as follows:

- In en bloc signaling, the switch waits for all digits of the called-party number from the user and then sends all the digits in a single SETUP message.
- In overlap signaling, the called-party digits are sent out as they are dialed from the user, instead of waiting for an interdigit timer to expire.

  Note: The interdigit timer starts when the user presses a digit key. The timer is restarted when the user presses the next digit key. Expiration of the timer indicates the end-of-dial (EOD).

In the H.323 network, dialed digits can be sent out or received in either en bloc (normal dialing) or overlap modes.

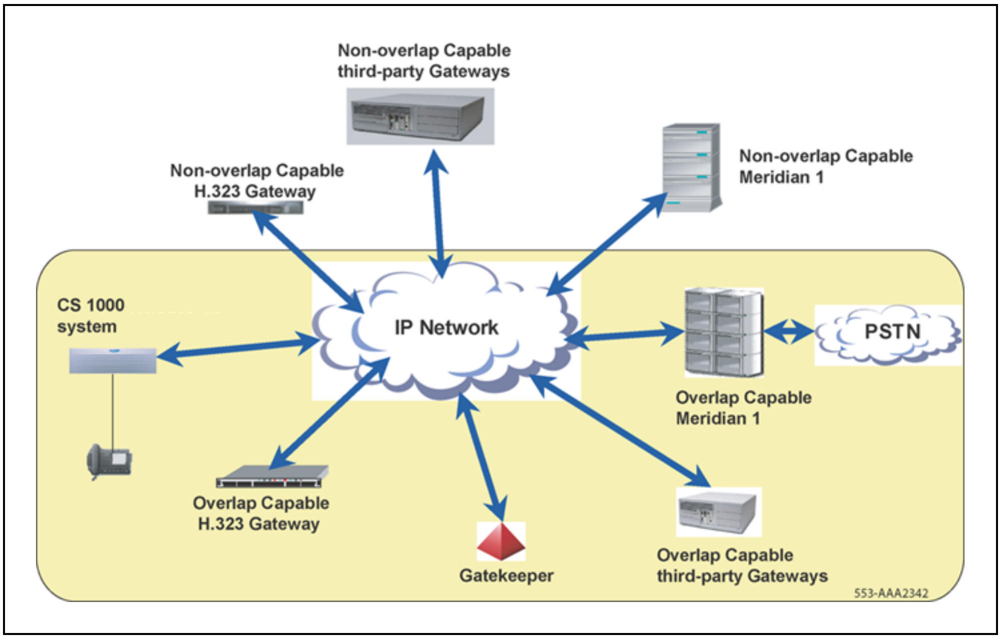
Overlap signaling consists of sending some digits of the called-party number in the first signaling message (SETUP messages) followed by further digits in subsequent signaling messages (INFORMATION messages).

Using the H.323 protocol and IP Peer Networking, overlap signaling is supported over IP between:

- two or more CS 1000 systems running CS 1000 Release 4.0 (or later) on both nodes
- CS 1000 IP Peer systems running CS 1000 Release 4.0 (or later) and another gateway (either a Nortel or third-party gateway) supporting overlap signaling (provided the capability is enabled on the gateway)

Figure 141 "Network diagram" (page 253) shows a network diagram with overlap signaling.

Figure 141
Network diagram


Non-overlap Capable third-party Gateways
Non-overlap Capable H.323 Gateway
Non-overlap Capable Meridian 1
CS 1000 system
IP Network
PSTN
Overlap Capable Meridian 1
Overlap Capable H.323 Gateway
Gatekeeper
Overlap Capable third-party Gateways
553-AAA2342

Advantages of overlap signaling

Overlap signaling allows the system to initiate a call from the originating node (towards the terminating node) while the originator is still dialing digits. As a result, overlap signaling improves the call setup time. Overlap signaling accelerates the transmittal of dialed digits which allows the terminating node to determine if the complete directory number (DN) is dialed. It also reduces the post-dial delay in networks where variable-length dialing plans are used.

Overlap signaling is useful when a system cannot determine the completion of all the digits, unless the caller terminates dialing with an octothorpe (#). For example, when a caller dials international numbers or when a caller dials private numbers where sub-DN digits may not be fully known across the whole network.

> ***Note:*** If overlap signaling is enabled on the Virtual Trunk D-channel for H.323, and the call is tandemed to DTI/Analog/DTI2, configure the Overlap Length parameter OVLL in the Route List Block for the DTI/Analog/DTI2 as 0.

Overlap signaling is in use in several countries with variable-length dialing plans (for example, Germany, Belgium, and Italy, and some other countries in Europe and Asia).

Overlap signaling also can improve interoperability with third-party gateways.

PSTN-destined calls

Overlap signaling support mainly impacts outgoing calls destined for PSTN terminations. Both line-originating calls and tandem trunk calls require overlap support.

This feature is applicable to PSTN calls with CS 1000 systems, because such calls can tandem through an IP Peer H.323 Gateway.

Feature capabilities

IP Peer Overlap Signaling includes the following capabilities:

- IP Peer overlap signaling support using the H.323 protocol
- Gatekeeper overlap signaling support
- Overlap sending/receiving configuration support
- Overlap signaling to en bloc conversion
- Tandem overlap signaling support

Overlap signaling support using the H.323 protocol

Overlap signaling is supported over IP Peer using version 4-compliant H.323 protocol signaling, as specified by the ITU-T H.323 and companion H.225 and H.245 standards.

IP Peer overlap signaling using H.323 is modeled on and parallels the Primary Rate Interface (PRI) overlap signaling. For more information on overlap signaling, refer to *ISDN Primary Rate Interface: Features* (NN43001-569-B1).

H.323 Gatekeeper overlap signaling support

The H.323 Gatekeeper provides support for overlap signaling.

When a CS 1000 H.323 Gatekeeper receives an ARQ message from the gateway, the message can include enough digits to resolve the address, or it can be incomplete (because overlap signaling has started but not completed). If it is incomplete (that is, the number is an incomplete prefix of one or more entries in the dialing plan), then the Gatekeeper supports overlap signaling by replying to the gateway with an "incomplete address" rejection reason.

The H.323 Gatekeeper also replies when the following occur:

- The number is invalid (that is, there are no possible matches in the dialing plan).
- There are at least two H.323 Gatekeepers in the network (one H.323 Gatekeeper that received the ARQ and could not resolve it, and a second H.323 Gatekeeper to receive the LRQ), and one of the following events occur:

- — at least one H.323 Gatekeeper failed to respond
- — the local H.323 Gatekeeper is provisioned with a default IP destination

The local Gatekeeper replies with an Admission Confirm (ACF) message. The ACF message includes the default IP destination and additional information. This additional information tells the gateway that the call handling has two options:

- The gateway can use the provided information and immediately continue with the call.
- The Gateway can carry out overlap to en bloc conversion and retry the ARQ.

For more information, refer to Appendix "H.323 Gatekeeper overlap signaling support" (page 327).

Overlap sending and receiving configuration support

Overlap sending and receiving are configurable for H.323 endpoints over IP Peer.

The user has the option to turn overlap sending and receiving on or off for the H.323 signaling gateway. In addition, the user can turn off overlap sending on specific destinations on an IP route (using the same signaling gateway) which is overlap enabled.

> ***Note:*** The IP Peer Overlap Signaling feature provides the ability to terminate overlap calls at an en bloc destination; however, this approach may not be efficient. If the nodes in the network are capable of supporting overlap signaling, then Nortel recommends that all nodes in the network be configured to use overlap signaling for optimal efficiency.

If a network must be configured such that some calls are en bloc and all other calls are overlap, then there are two ways to configure the network to avoid overlap to en bloc conversion. The two methods are:

- Configure separate Route List Index (RLI) instances to create different Route List Blocks (RLB). This is the preferred method.
- Configure separate Signaling Servers for en bloc and overlap traffic.

Separate Route List Index (RLI) instances

Nortel recommends that separate Route List Index (RLI) instances be configured to create different Route List Blocks (RLB) for en bloc and overlap traffic to the same CS 1000 Signaling Server. Using different RLBs for overlap and en bloc calls saves provisioning and hardware resources, because only one D-channel on the Call Server and one Signaling Server are used.

Note: RLBs provide an option to configure the Overlap Length (OVLL) for different RLIs. If OVLL is defined as 0, then (for any route on that particular RLI) all the calls made over that route are en bloc.

Separate Signaling Servers

As an alternate approach, separate CS 1000 Signaling Servers can be used for en bloc and overlap traffic.

Two CS 1000 Signaling Servers can be configured, where:

- one Signaling Server carries overlap signaling traffic
- one Signaling Server carries en bloc traffic

This configuration requires two D-channels on the Call Server. One D-channel can be configured as en bloc and the other as overlap.

Overlap to en bloc conversion

Nortel recommends that all nodes in the network that are capable of overlap signaling have overlap receiving enabled as a minimum, and, if possible, have both overlap receiving and overlap sending enabled.

However, a network can have nodes that are not capable of supporting overlap signaling. If an H.323 overlap call encounters such a destination, then the originating node can complete the call by reverting to en bloc mode. This is known as overlap to en bloc conversion.

The following two events can occur when an H.323 SETUP message (for an overlap-capable call) reaches an en bloc destination:

- In response to the SETUP message, an H.323 CALL PROCEEDING message is sent indicating the end-of-dial. This message is followed by a call clear, which indicates an incomplete number may occur.
- The call can clear immediately, indicating an incomplete number.

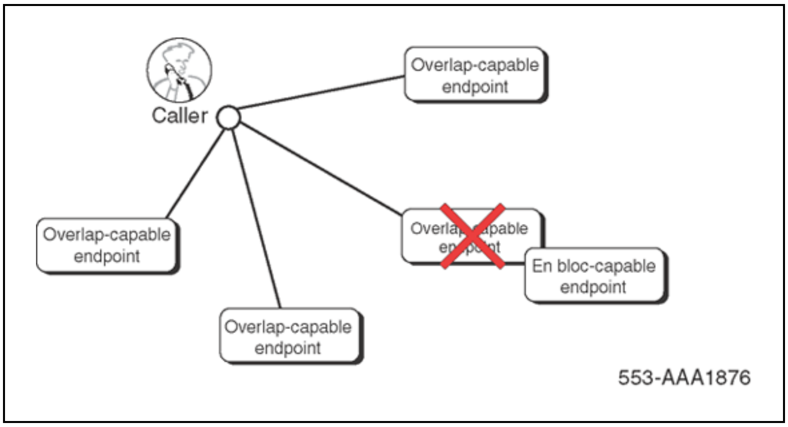
In both cases, overlap to en bloc conversion begins. The interdigit timer starts and digits are collected until an end-of-dial indication. That is, the interdigit timer expires on the Call Server, triggering the end-of-dial indication or the Call Server sends an end-of-dial indication for some other reason; this mechanism exists within the Call Server messages. The reasons can include reaching the provisioned maximum length, user input, or a tandem transmission of the end-of-dial indication. At that time, the gateway sends a new H.323 SETUP message with all received digits, and an end-of-dial indication. All further call processing occurs using en bloc signaling.

Changing the provisioning from using overlap signaling (to reach a destination) to using en bloc signaling

Figure 142 "Changing an overlap-capable endpoint to an en bloc endpoint" (page 257) shows a network of overlap-capable endpoints where one of the endpoints must be changed to en bloc-capable.

> ***Note:*** "Overlap-capable endpoint" implies that signaling to this destination uses overlap dialing, while "En bloc-capable endpoint" implies that overlap signaling is not used to reach this destination. The true capabilities of the destinations are not known at the originator.'

Figure 142
Changing an overlap-capable endpoint to an en bloc endpoint



For efficiency, configure another RLI as en bloc in LD 86 to change that endpoint from overlap signaling-capable to en bloc:

1. In LD 86, define a new RLI.
2. Configure the Overlap Length (OVLL) prompt to 0.

   Note: The OVLL prompt determines the number of digits required before the SETUP message is sent. If OVLL = 0, then all the dialed digits are sent in the SETUP message and the call is an en bloc call (even if LD 17 is configured for overlap signaling).

3. Change the entries pointing to the destination (that just changed to en bloc) to use the new RLI. After all ESN and CDP code entries have been changed, you can then remove the overlap RLI.

Example: Assume that Location Code (LOC) 425 currently uses the overlap-capable RLI 21 to call an overlap node. If that node changes to en bloc, then the following changes must be made:

- In LD 86, define a new RLI (such as RLI 22) with OVLL configured to 0. (All other prompts in the RLI can be identical to the original RLI 21.)
- In LD 90, change the LOC 425 to use the new RLI 22.

Tandem overlap signaling support

In addition to supporting originating and terminating overlap calls, IP Peer Overlap Signaling also supports the following tandem scenarios:

- ISDN (en bloc/overlap) to IP Peer (H.323-overlap/en bloc)
- Non-ISDN (en bloc/overlap) to IP Peer (H.323-overlap/en bloc)
- IP Peer (H.323-overlap) to IP Peer (H.323-overlap/en bloc)
- IP Peer (H.323-overlap) to IP Peer (SIP)

Overlap signaling call flow

Any messaging after the Alerting message is identical to the en bloc call flow and is not repeated in this section.

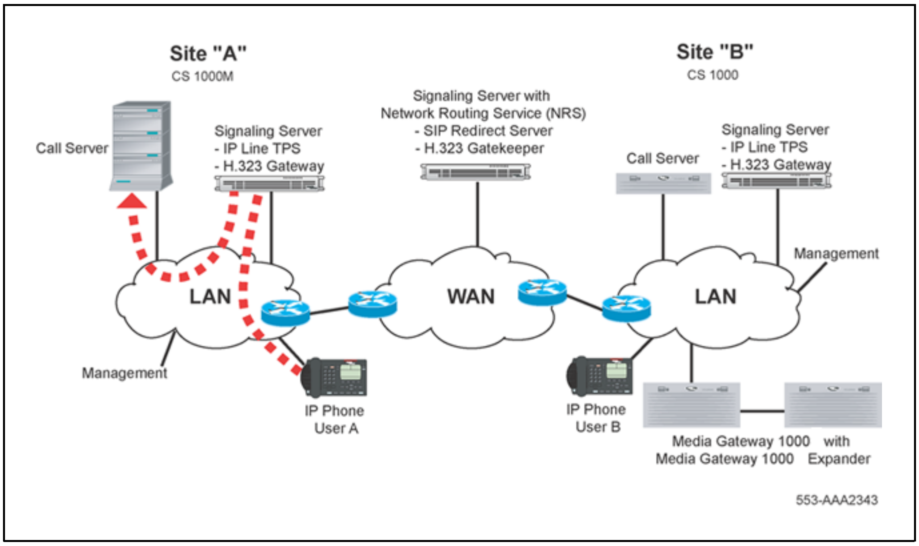
> ***Note:*** Only the primary messages are illustrated in the following call flows.

The following scenario describes the Direct IP Media Path functionality for a basic network call using overlap signaling:

Step **Action**

1 User A on Call Server A dials the DN of User B on Call Server B. Call Server A collects the routing-prefix digits through the Terminal Proxy Server (TPS) on Signaling Server A. See Figure 143 "User A dials User B" (page 258).

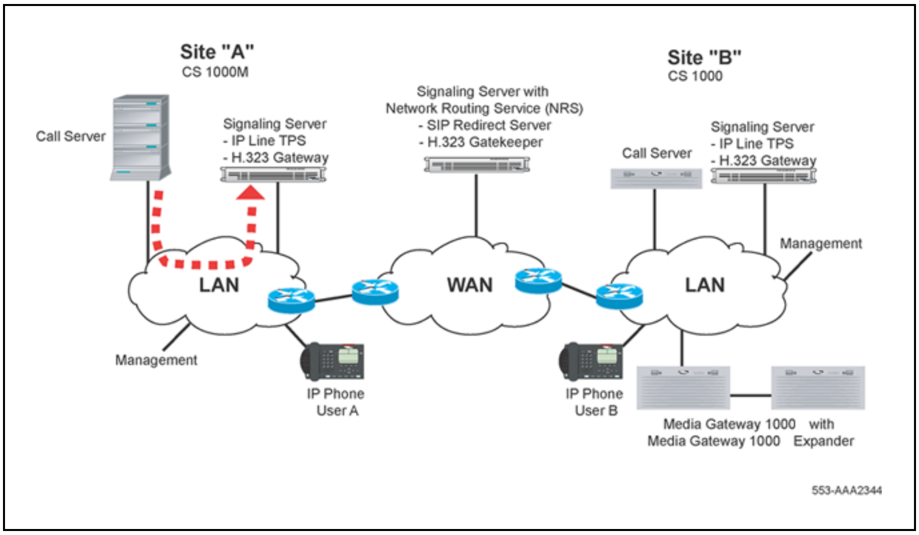
Figure 143
User A dials User B



2 Call Server A determines that the dialed DN is at another site reachable using overlap signaling. Call Server A selects the codec list, allocates bandwidth, and routes the call to the IP network using a Virtual Trunk and an H.323 Gateway. See Figure 144 "Call Server A routes the call to the IP network" (page 259).

> ***Note:*** To select which Virtual Trunk to use for routing, Call Server A examines the number dialed and uses various trunk routing and signaling features (for example, ESN and MCDN).

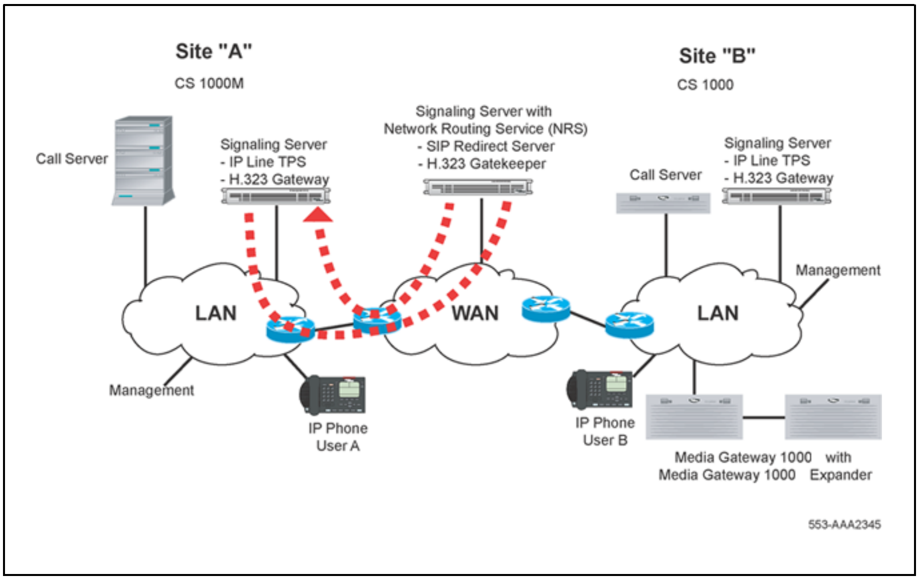
Figure 144
Call Server A routes the call to the IP network



3 H.323 Gateway A asks the NRS (specifically the H.323 Gatekeeper) to search for the dialed DN in the database (for example, within the appropriate CDP domain). If the NRS (H.323 Gatekeeper) can unambiguously resolve the destination digits, it sends the IP address of H.323 Gateway B to H.323 Gateway A. See Figure 145 "The H.323 Gatekeeper sends the IP address of H.323 Gateway B to H.323 Gateway A" (page 260).

Otherwise, the NRS requests more digits.

Figure 145
The H.323 Gatekeeper sends the IP address of H.323 Gateway B to H.323 Gateway A



4 User A dials an additional digit. The TPS forwards it to Call Server A. See Figure 143 "User A dials User B" (page 258).

5 Call Server A forwards the digits to H.323 Gateway A on the Signaling Server. See Figure 144 "Call Server A routes the call to the IP network" (page 259).

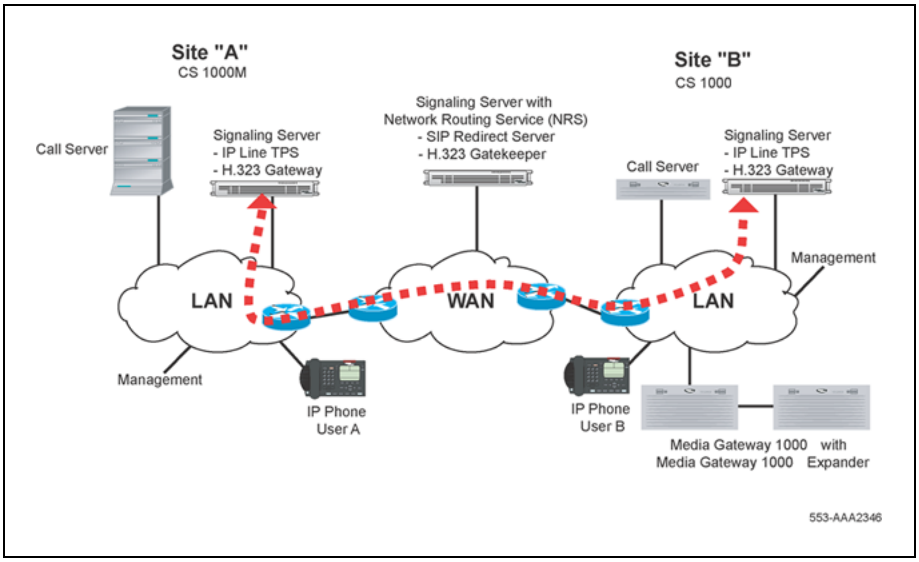
6 H.323 Gateway A asks the NRS (specifically the H.323 Gatekeeper) to search for the dialed DN in the database (for example, within the appropriate CDP domain). If the NRS (H.323 Gatekeeper) can unambiguously resolve the destination digits, it sends the IP address of H.323 Gateway B to H.323 Gateway A. See Figure 145 "The H.323 Gatekeeper sends the IP address of H.323 Gateway B to H.323 Gateway A" (page 260).

Otherwise, the NRS requests more digits.

> ***Note:*** Until the call succeeds, step 4, step 5, and step 6 are repeated for each new dialed digit.

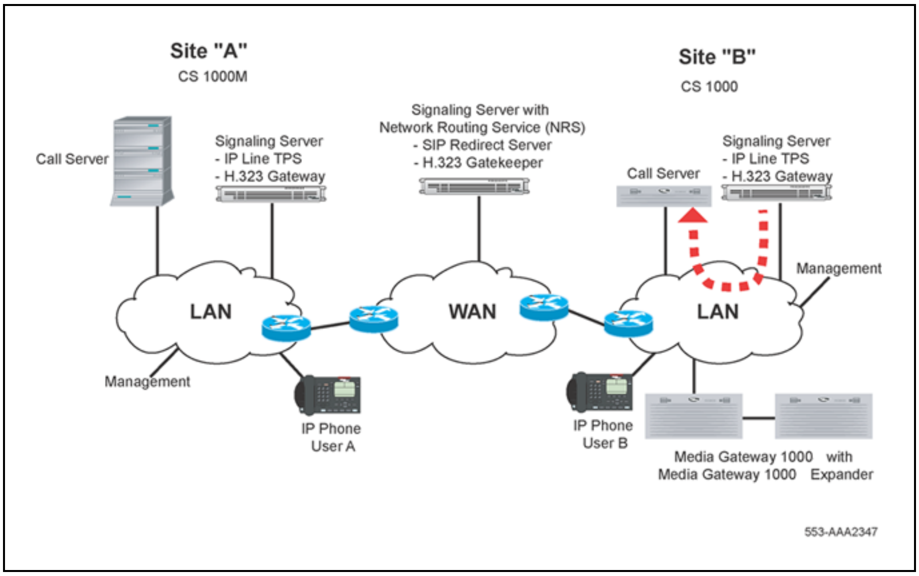
7 H.323 Gateway A sends a SETUP message to H.323 Gateway B, including the DN information and an indication that H.323 Gateway A is overlap capable. H.323 Gateway B replies with a SETUP ACK indicating that it is also overlap capable. See Figure 146 "H.323 Gateway A sends a SETUP message to H.323 Gateway B" (page 261).

Figure 146
H.323 Gateway A sends a SETUP message to H.323 Gateway B



8 H.323 Gateway B treats the call as an incoming overlap signaling call from a Virtual Trunk. H.323 Gateway B sends the call to Call Server B over a Virtual Trunk. Call Server B also treats the call as an incoming call from a Virtual Trunk. See Figure 147 "Gateway B sends the call to Call Server B over a Virtual Trunk" (page 261).

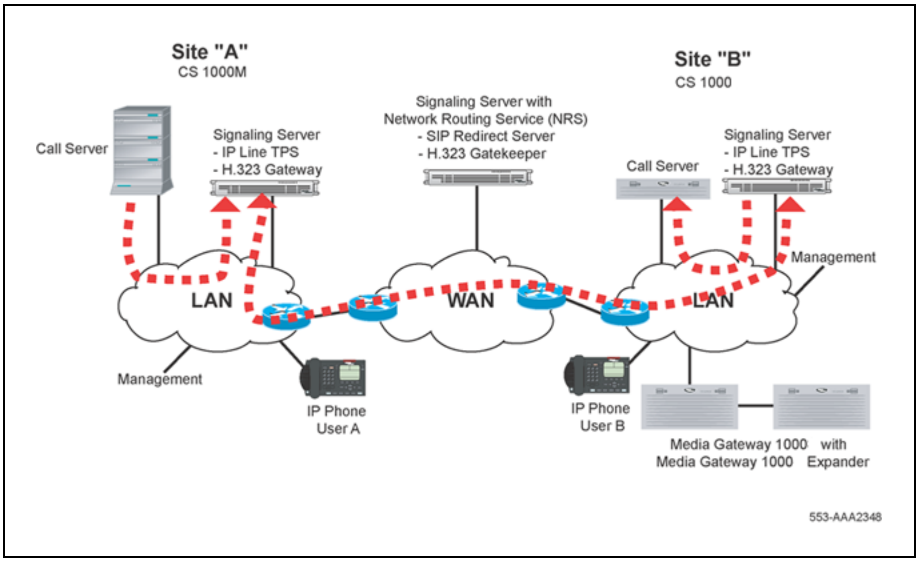
Figure 147
Gateway B sends the call to Call Server B over a Virtual Trunk



9 User A on Call Server A dials additional digits. See Figure 143 "User A dials User B" (page 258).

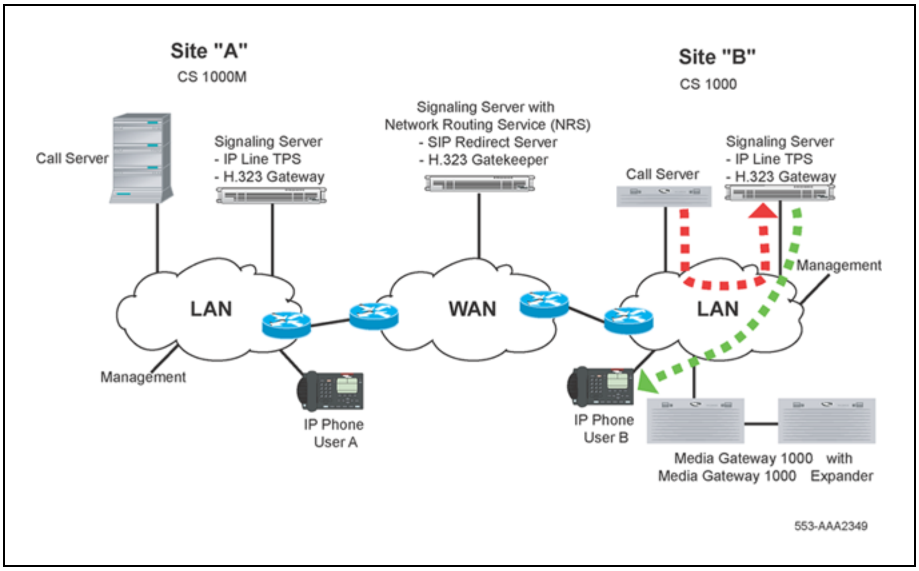
10 Call Server A sends the new digits to Call Server B through the two gateways. This repeats until Call Server B receives all the digits. At that time, Call Server B sends an end-of-dial indication to Call Server A. See Figure 148 "Call Server A sends digits to Call Server B" (page 262).

Figure 148
Call Server A sends digits to Call Server B



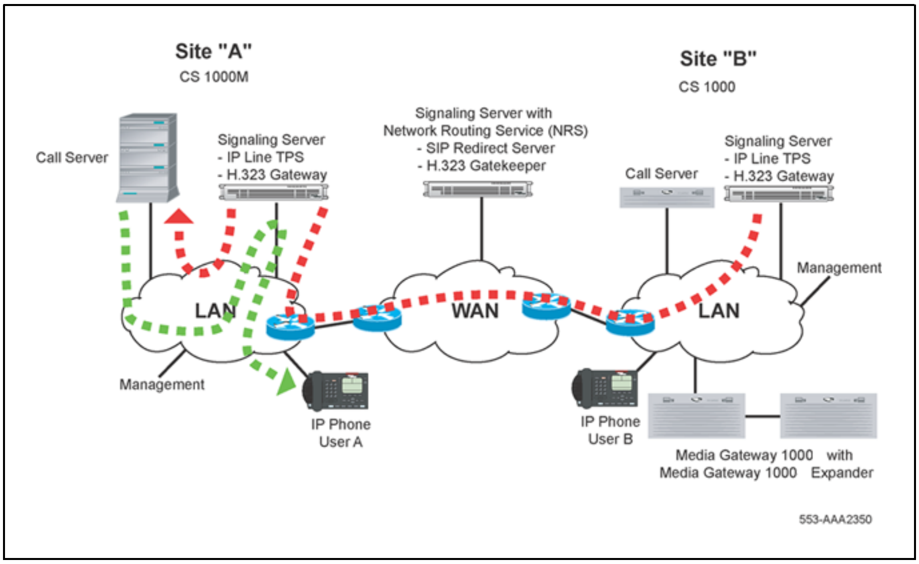
11 Call Server B selects the codec, allocates bandwidth, rings the telephone, and sends an alerting message to H.323 Gateway B. See Figure 149 "Call Server B sends an alerting message to H.323 Gateway B" (page 262).

Figure 149
Call Server B sends an alerting message to H.323 Gateway B



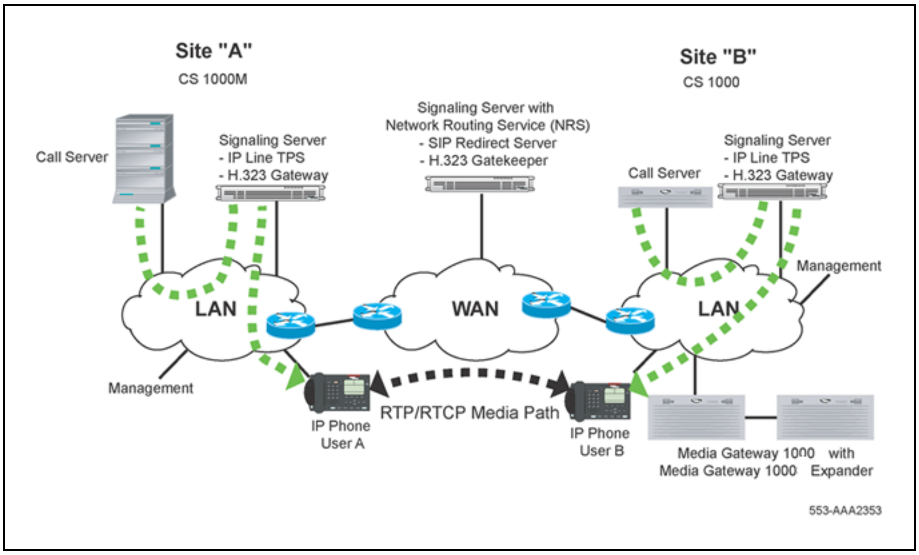
12 H.323 Gateway B sends an alerting message to Call Server A. Call Server A requests that the IP Phone play ringback tone. See Figure 150 "H.323 Gateway B sends an alerting message to Call Server A" (page 263).

Figure 150
H.323 Gateway B sends an alerting message to Call Server A



13 User B answers the call. A message is sent to Call Server B through the TPS on the Signaling Server. See Figure 151 "User B answers the call" (page 263).

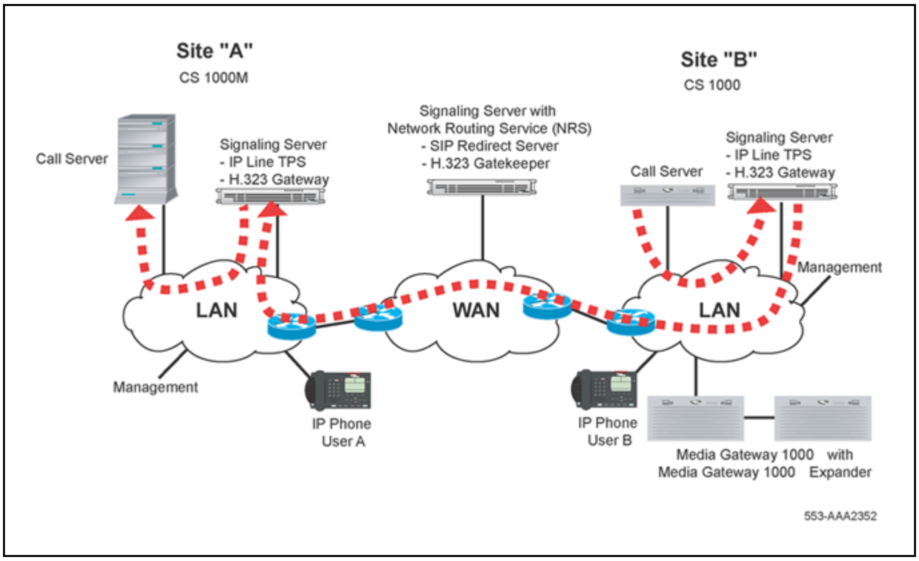
Figure 151
User B answers the call



14 Call Server B sends a CONNECT message to H.323 Gateway B. H.323 Gateway B sends an H.323 CONNECT message to H.323

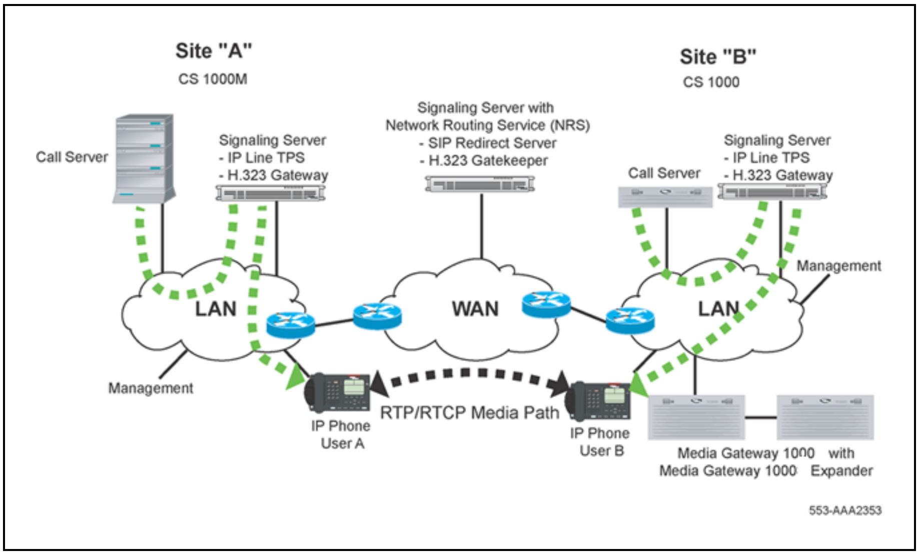
Gateway A. H.323 Gateway A forwards the message to Call Server A. See Figure 152 "Call Server B sends a CONNECT message to Gateway B and onto Call Server A" (page 264).

Figure 152
Call Server B sends a CONNECT message to Gateway B and onto Call Server A



15 The Call Servers tell the IP Phones to start the direct IP media paths. The IP Phones then begin to transmit and receive voice over the IP network. See Figure 153 "IP Phones start the direct media paths" (page 264).

Figure 153
IP Phones start the direct media paths



—End—

Feature packaging

IP Peer Overlap Signaling requires the following packages:

- Overlap Signaling (OVLP) package 184
- H.323 Virtual Trunk (H323_VTRK) package 399

Note: The packaging for H.323 includes the Overlap Signaling package.

Configuring overlap signaling on the Call Server

The following task summary list includes all the tasks required to configure IP Peer with overlap signaling. In particular, overlap signaling is configured using LD 17 and LD 86 as follows:

- Use LD 17 to configure the D-channel to support overlap signaling.
- Use LD 86 to configure the number of digits to be included in the SETUP message.

When configuring overlap signaling, the network must be optimized using a combination of the Overlap Length (OVLL) prompt in LD 86 and the Overlap Timer (OVLT) prompt in LD 17. Nortel recommends that:

- OVLL be configured to a reasonable length such that the Gatekeeper can resolve the called-party number with a minimum number of transactions
- OVLT be configured to 1 second

WARNING

When using SPNs to provide local, national, and international number handling, the incoming local, national, and international numbers are treated as en bloc. If the number was sent using overlap signaling, the call will always be incomplete.

If the Call Server receives an unknown type (CDP, LOC, or SPN) on an overlap capable D-channel, the Call Server processes the call as overlap. If it receives E.164 numbers, the Call Server treats them as North American formatted (and therefore, en bloc).

Task summary list

The following is a summary of the tasks in this section:

1. LD 17 – Configure D-channels to support overlap signaling.
2. LD 16 – Configure the H.323 route.
3. LD 86 – Configure the Route List Block for the Virtual Trunk route and configure the minimum number of digits included in the overlap signaling SETUP message.

4. LD 87 – Configure the CDP steering codes.
5. LD 90 – Configure E.164 plan call types and private plan Location Codes.
6. LD 90 – Configure Special Numbers.

   Note: Only the Overlays directly affected by the overlap signaling feature are included here.

LD 17 Configure D-channels to support overlap signaling.

Prompt	Response	Description
REQ	chg	Change existing data
TYPE	adan	Action Device And Number
ADAN	new dch xx	Action Device And Number, where xx is 0-63.
CTYP	 dcip	Card Type D-channel over IP
BANR	YES	Enable security banner printing option
IFC	sl1	Interface type for D-channel
H.323		Indicates overlap signaling prompts for H.323
OVLR	yes	Overlap Receiving
OVLS	yes	Overlap Sending
OVLT	0-(1)-8	Overlap Timer (in seconds) The timer controls the interval between the sending of INFORMATION messages. Defaults to 1 for D-channel over IP ***Note:*** OVLT applies only to Overlap Sending (OVLS = YES).

In the Route Data Block, the zone parameter makes the codec selections and calculates the bandwidth usage for calls to the trunk members of a given route.

LD 16 Configure the H.323 route.

Prompt	Response	Description
REQ	new	Add a new route.
TYPE	rdb	Route Data Block
CUST	xx	Customer number as defined in LD 15.

Prompt	Response	Description
ROUT		Route number
	0-511	Range for Large System and CS 1000E system
	0-127	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T
DES	x...x	Designator
		The designator field for the trunk groups. This designator can be 0-16 alphanumeric characters.
TKTP VTRK	tie yes	Trunk Type TIE trunk Virtual Trunk route, where: YES = This route is for Virtual Trunk NO = This route is not for Virtual Trunk (default)
ZONE	0-255	Zone for codec selection and bandwidth management
PCID	H323	Protocol ID for the H.323 route
		Defines the route as an H.323 route.
NODE	xxxx	Node ID
		Where the Node ID matches the node of the Signaling Server. The Node ID can have a maximum of four numeric characters.
ISDN	yes	Integrated Services Digital Network option
MODE		Mode of operation
	ISLD	Route uses ISDN Signaling Link (ISL)
		ISLD is allowed only if ISDN = YES, and the Integrated Services Digital Network Signaling Link (ISL) package 147 is equipped. ISLD is allowed only on ISA and TIE trunks.
DCH	0-159	D-channel number
IFC	sl1	Interface type for route (IFC responses are listed in *Software Input/Output: Administration* [NN43001-611])
SRVC	a...a	Service type for AT&T ESS connections (SRVC responses are listed in *Software Input/Output: Administration* [NN43001-611])
PNI	(0)-32700	Private Network Identifier
NCNA	(YES)	Network Calling Name Allowed
NCRD	YES	Network Call Redirection

Prompt	Response	Description
INAC	(NO) YES	Insert ESN Access Code Inserts the ESN access code in an incoming private network call. INAC enables an ESN access code to be automatically added to an incoming ESN call from a private network. If INAC = YES, then digit insertion (INST) for NARS or BARS calls is bypassed and Access Code 1 (AC1) is used for all call types. However, calls can be specifically defined to use Access Code 2 (AC2) in LD 15 at the AC2 prompt. INAC is prompted when the route type is either a TIE trunk or an IDA trunk with DPNSS1 signaling.
ICOG		Incoming and Outgoing trunk.
	IAO	Incoming and Outgoing
ACOD	x...x	Access Code for the trunk route.

Nortel recommends that all routes in a Route List Block (RLI) be configured as either overlap or en bloc. That is, an en bloc route should not have alternate routes that are configured as overlap, and vice versa. Erratic behavior can occur when overlap and en bloc routes are configured as alternate routes. Normal behavior occurs on alternate routes as long as the alternate route has the same overlap capabilities as the main route.

A warning message is displayed if alternate routes are configured as a different type from the main route.

LD 86 Configure the Route List Block for the Virtual Trunk route and configure the minimum number of digits included in the SETUP message.

Prompt	Response	Description
REQ	NEW	Create new data block
CUST	xx	Customer number as defined in LD 15.
FEAT	RLB	Route list block
...		
RLI		Route List Index to be accessed
	0-127 0-255 0-999	CDP and BARS NARS FNP

Prompt	Response	Description
ENTR	xxx	Entry number for NARS/BARS Route list Where xxx = • 0-63 Entry number for NARS/BARS Route List • 0-6 Route list entry number for CDP • X Precede with x to remove
ROUT		Route number
	0-511	Range for Large System and CS 1000E system
	0-127	Range for Small System, Media Gateway 1000B, and Media Gateway 1000T ***Note:*** The route must be overlap capable.
...		
ENTR	<cr>	Entry number for NARS/BARS Route list
ISET	(0)-8	Initial Set Number of entries in Initial Set for route list block.
...		
OVLL	(0)-24	Overlap Length Number of digits required before the SETUP message is sent. If OVLL = 0 then all the dialed digits are sent in a single SETUP message and the call is an en bloc call (even if LD 17 suggests overlap signaling). A value of x, where x is a 1 to 24, that x digits are required before sending the SETUP message.

Prompt	Response	Description
		Note: Setting the OVLL to the expected digit string length (for example, OVLL = 7 when using seven-digit UDP) effectively forces en bloc. The SETUP message must have all seven digits before the message is sent. Therefore, the whole number is sent in the first message.

LD 87 Configure the CDP steering codes.

Prompt	Response	Description
REQ	NEW	Add new data.
CUST	xx	Customer number as defined in LD 15.
FEAT	CDP	Coordinated Dialing Plan
TYPE	 DSC TSC	Type of steering code Distant Steering Code Trunk Steering Code
DSC	x..x	Distant Steering Code Up to 4 digits; up to 7 digits with Directory Number Expansion (DNXP) package 150.
- FLEN	(0)-10	Flexible Length number of digits ***Note:*** See "Flexible Length number of digits implications" (page 275) for more information about FLEN.
TSC	x..x	Trunk Steering Code Up to 4 digits, up to 7 digits with Directory Number Expansion (DNXP) package 150.
- FLEN	(0)-24	Flexible Length number of digits ***Note:*** See "Flexible Length number of digits implications" (page 275) for more information about FLEN.

LD 90 Configure E.164 plan call types and private plan Location Codes.

Prompt	Response	Description
REQ	NEW CHG	Create new data block Change existing data block
CUST	xx	Customer number as defined in LD 15.
FEAT	NET	Network Translator (Network translation tables)
TRAN	 AC1 AC2	Translator Access Code 1 (NARS/BARS) Access Code 2 (NARS)

Prompt	Response	Description
TYPE	 LOC	Type Location Code
LOC	xxx y..y	Location code, where xxx = home location code and y..y = extended code of 1-4 digits. The extended code is optional.
- FLEN	(0)-10	Flexible Length number of digits Enter the maximum number of digits expected. When this number of digits is dialed, dialing is considered to be complete and end-of-dial processing begins. Default is zero (0) digits. ***Note:*** See "Flexible Length number of digits implications" (page 275) for more information about FLEN.

LD 90 Configure Special Numbers.

Prompt	Response	Description
REQ	NEW CHG	Create new data block Change existing data block
CUST	xx	Customer number as defined in LD 15.
FEAT	NET	Network Translator (Network translation tables)
TRAN	 AC1 AC2	Translator Access Code 1 (NARS/BARS) Access Code 2 (NARS)
TYPE	 SPN	Type Special Number Translation
SPN	xxx	Special Number translation Enter the SPN digits in groups of 3 or 4 digits, separated by a space (e.g., xxxx xxx xxxx). The SPN can be up to 19 digits long. The maximum length no longer depends on whether or not the first digit of the SPN is a "1". That restriction has been removed. The maximum number of groups allowed is 5.
- FLEN	(0)-24	Flexible Length number of digits Enter the maximum number of digits expected. When this number of digits is dialed, dialing is considered to be complete and end-of-dial processing begins.

Prompt	Response	Description
		Default is zero (0) digits. ***Note:*** See "Flexible Length number of digits implications" (page 275) for more information about FLEN.

Configuring overlap signaling using Element Manager

To configure a D-channel to support overlap signaling, follow the steps in Procedure 25 "Configuring D-channels to support overlap signaling" (page 272).

Procedure 25
Configuring D-channels to support overlap signaling

Step Action

1 Log in to CS 1000 Element Manager.

2 Select **Routes and Trunks > D-Channels** from the EM Navigator.

The **D-Channel** web page opens.

3 Click the **Edit** button associated with the D-channel.

The **D-Channel xx Property Configuration** web page opens where xx is the D-channel number.

4 Choose **Advance Options (ADVOPT)**.

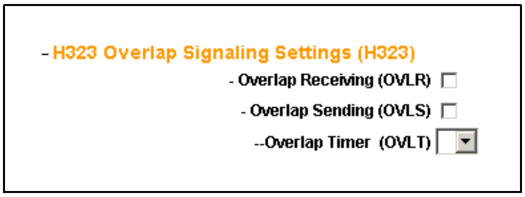
5 Choose **H.323 Overlap Signaling Settings (H323)** (see Figure 154 "H.323 Overlap Signaling" (page 272)).

a. Select the **Overlap Receiving (OVLR)** check box.

b. Select the **Overlap Sending (OVLS)** check box.

c. Select a timer value (in seconds) from the **Overlap Timer (OVLT)** drop-down list.

Figure 154
H.323 Overlap Signaling



- H323 Overlap Signaling Settings (H323)

- Overlap Receiving (OVLR) □

- Overlap Sending (OVLS) □

--Overlap Timer (OVLT) ▾

6 Click **Submit**.

—End—

To configure the number of digits required before the SETUP message is sent, follow the steps in Procedure 26 "Configuring the minimum number of digits included in the SETUP message" (page 273).

Procedure 26
Configuring the minimum number of digits included in the SETUP message

Step	Action
1	Refer to Procedure 13 "Configuring the Route List Block" (page 202).
2	Enter a value for **Overlap Length (OVLL)**. If OVLL = 0 (the default), then all the dialed digits are sent in a single SETUP message and the call is an en bloc call (even if Procedure 25 "Configuring D-channels to support overlap signaling" (page 272)/LD 17 suggests overlap signaling). A value of x, where x is 1-24, indicates that x digits are required before sending the SETUP message. ***Note:*** Setting the OVLL to the expected digit string length (for example, OVLL = 7 when using seven-digit UDP) effectively forces en bloc. The SETUP message must have all seven digits before the message is sent. Therefore, the whole number is sent in the first message.

—End—

Overlay changes for overlap signaling

LD 17 displays the H323 prompt is displayed for a D-channel over IP (type DCIP). The H323 prompt has three key prompts, OVLR, OVLS, and OVLT, that are provided for an H.323 D-channel.

> ***Note:*** This prompt sequence is displayed only for a D-channel of type DCIP if the H.323 Virtual Trunk (H323_VTRK) package 399 and Overlap Signaling (OVLP) package 184 are enabled. Otherwise, the OVLR, DIDD, OVLS, and OVLT prompt sequence is displayed.

The user must configure OVLS, OVLR, and OVLT in LD 17 in order for overlap signaling to work.

> ***Note:*** The D-channel must be disabled before modifying the OVLR and OVLS prompts. The OVLR and OVLS data must be transmitted to the Signaling Server. This occurs only when the D-channel is enabled.

- If the Call Server is to send overlap calls over IP, then Overlap Sending (OVLS) must be configured as YES. This setting turns on overlap sending from the Call Server to the IP domain.
- If the Call Server is to receive overlap calls over IP, then Overlap Receiving (OVLR) must be configured as YES. This setting turns on overlap signaling from the IP domain to the Call Server.
- The Overlap Timer (OVLT) prompt only has meaning for Overlap Sending (OVLS = YES). The OVLT value indicates the time the system waits to accumulate digits to send in an INFORMATION message after the SETUP message is sent. The valid values for OVLT are 0-8 where:
  - — A value of 0 results in the generation of an INFORMATION message for every digit dialed after the minimum overlap called number length (as provisioned in LD 86 for the RLI).
  - — A value of 1 is the default value for a D-channel over IP.

In LD 86, a warning is issued if a mixture of IP capable overlap routes and en bloc capable routes exist in an RLI. The warning is also issued if an en bloc IP route coexists with overlap capable routes. The warning is displayed only at the Call Server login window. It is not transmitted to Element Manager.

The use of the Flexible Length number of digits (FLEN) prompt has changed (in LD 87 and LD 90) for overlap signaling but has not changed for en bloc.

With IP Peer overlap signaling calls, the usage of the FLEN prompt is changed as follows:

- If FLEN = 0, then (in general) overlap handling has not changed. The SETUP message is sent once the OVLL digits are received and the dialing plan entry can be determined. However, the end-of-dial timer starts, and on expiration, the Call Server sends an INFORMATION message with Sending Complete to indicate end-of-dial.
- If FLEN is greater than 0 and also greater than both of the following:
  - — the length of the digit string provisioned in LD 87 or LD 90, and
  - — the OVLL value,

  then overlap signaling meets the two requirements for the SETUP message. After that, further digits are sent in the INFORMATION messages. In addition, for IP overlap signaling, when the value configured for FLEN is reached, the INFORMATION message carrying the digits also carries the Sending Complete Information Element (IE).

  Note: An IE is a unit of information in Q.931 and H.323 messaging.

- If FLEN is less than OVLL, then the SETUP message is sent immediately. To ensure that the Signaling Server does not wait for more digits, the SETUP message also includes the Sending Complete IE.

With en bloc calls, the usage of the FLEN prompt is as follows:

- If FLEN is a non-zero value, then the Call Server collects digits until the total count of collected digits equals FLEN. The Call Server then sends a SETUP message.
- If FLEN = 0, then the Call Server uses an end-of-dial timer to determine when it has a completed number. The Call Server collects digits, restarting the end-of-dial timer after each digit, and waits for the timer to expire to send the SETUP message.

Flexible Length number of digits implications

A non-zero FLEN value indicates the number of digits the system should expect for the current number type and plan entry. When the digits collected reach the expected length, the system sends an end-of-dial indication to the remote switch. A value of 0 means the length is unknown and FLEN = 0 has a specific impact on the system.

En bloc dialing handles an unknown length by using an end-of-dial timer. It uses the end-of-dial timer to decide how many digits it must collect. When the timer expires, all received digits are sent in the SETUP message.

When PRI uses overlap signaling and FLEN = 0, the network relies on the remote switch to determine the correct length. The originating switch can use overlap signaling to send the digits once the OVLL and dialing plan entry requirements are met.

For IP, however, the remote switch may be one of many devices (for example, a CS 1000 system, an H.323 gateway, or a Business Communications Manager (BCM) node). The remote switch may also be overlap- or en bloc-capable. An overlap call to an overlap destination is not an issue. However, an en bloc destination requires overlap-to-en bloc conversion, which in turn requires knowledge of when a digit string is completed. Therefore, for overlap signaling on IP Peer to perform overlap-to-en bloc conversion with a FLEN of 0, the system must know when the digit string ended. As a result, unlike the PRI overlap-signaling case, when the end-of-dial timer expires (for an IP overlap signaling call) the Call Server sends a Sending Complete IE in the INFORMATION message to indicate end-of-dial.

Nortel recommends that all numbers with a known length set FLEN equal to the length of the digit string. For example, if all Location Codes (LOC) are eight digits in length, then use FLEN = 8 for all LOC codes. However, when the destination is unknown, use FLEN = 0. This process provides full overlap capability to an overlap-enabled destination, while providing the end-of-dial indication to allow interworking with an en bloc destination.

> ***Note:*** Dialing the octothorpe (#) forces an immediate end-of-dial, so the Call Server immediately provides end-of-dial treatment.

System log messages

The Signaling Server has a log file. A system log message is issued to this log file when the Signaling Server detects an incoming H.323 overlap signaling call that must revert to en bloc.

This system log message is output no more than once every hour. The message indicates the number of occurrences of overlap-to-en bloc conversion since the last system log message. No output is generated during a period in which no overlap-to-en bloc conversion occurred.

IP Peer interworking

Contents

This section contains information on the following topics:

Nortel products interworking

CS 1000M System interworking

A CS 1000M System internetworks with other Nortel products. This chapter discusses interworking between CS 1000M systems and the following products:

- Meridian 1 IE (IP Trunk Release 3.0 or later)
- Succession Release 3.0
- Business Communications Manager (BCM) Release 3.01 (or later)

Business Communications Manager

Business Communications Manager (BCM) can be part of an overall CS 1000 network. BCM can interwork with the H.323 Gatekeeper, supporting the basic network numbering plan and providing MCDN non-call associated signaling (such as Message Waiting Indication for network voice mail service).

> ***Note:*** IP Peer Networking with CS 1000 requires BCM Release 3.0 or higher.

Meridian 1 IE (IP Trunk Release 3.0 or later) / Succession 3.0

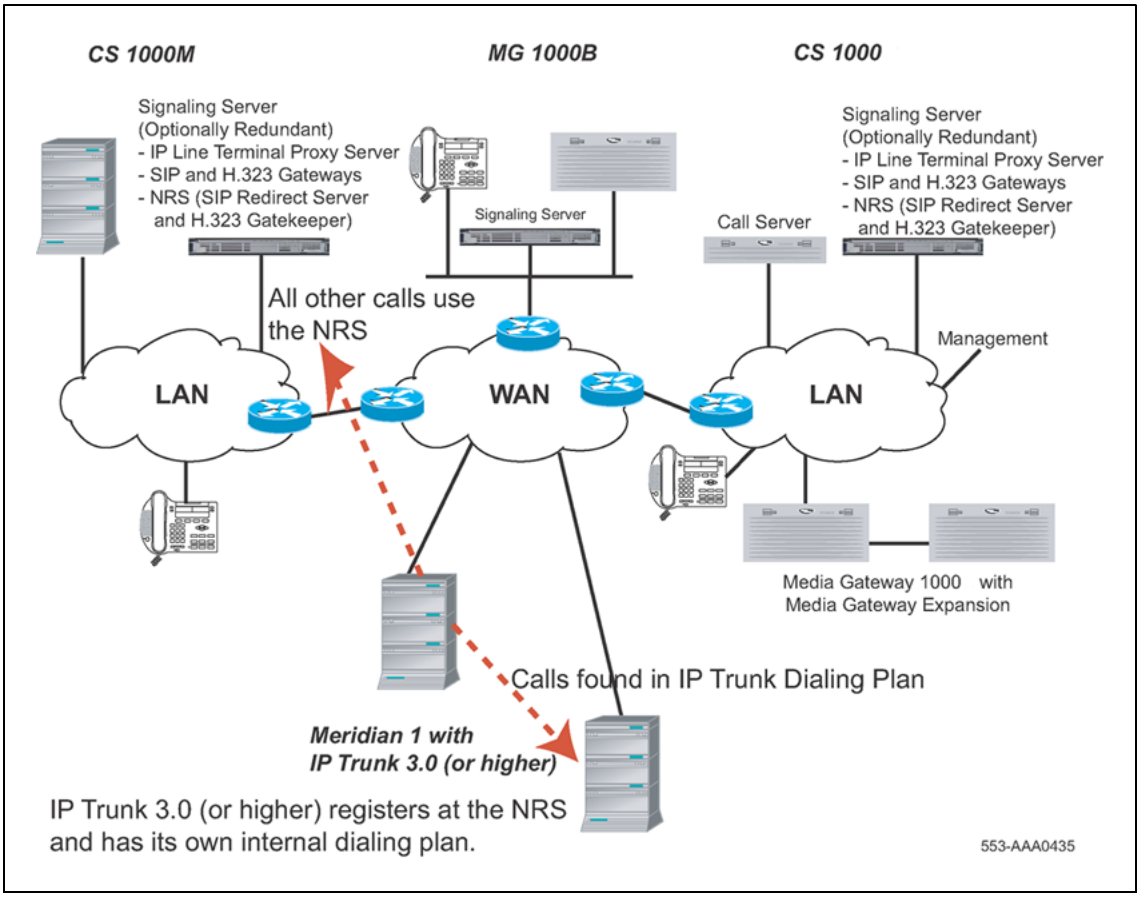
CS 1000 Release 4.0 and later networks with Meridian 1 Release 25.xx (or later) and Succession 3.0. Nortel Meridian Customer Defined Network (MCDN) protocol over PRI trunks provides the rich feature set currently available to networks of Meridian 1 Systems.

Any existing IP Trunks in the system must be upgraded to IP Trunk 3.0 (or later) in order to interwork with an IP Peer Networking node.

IP Peer Networking interworks with IP Trunk 3.0 (or later). It also supports all the MCDN features that IP Trunk 3.0 (or later) supports including Trunk Route Optimization.

With IP Trunk, the numbering plan is configured for each site. With IP Peer Networking, the NRS maintains the numbering plan for all sites. IP Trunk 3.0 (or later) maintains a point-to-point configuration. If a call is routed using IP Trunk 3.0 (or later) and the path is found, then the session is established. If the route path is not found, the lookup process is handed off to the NRS to resolve the route path. See Figure 155 "IP Peer to Meridian 1 IP Trunk 3.0 (or later) Interworking" (page 279).

Figure 155
IP Peer to Meridian 1 IP Trunk 3.0 (or later) Interworking



For a CS 1000M System to interwork with a Meridian 1 IE system, the following requirements must be met:

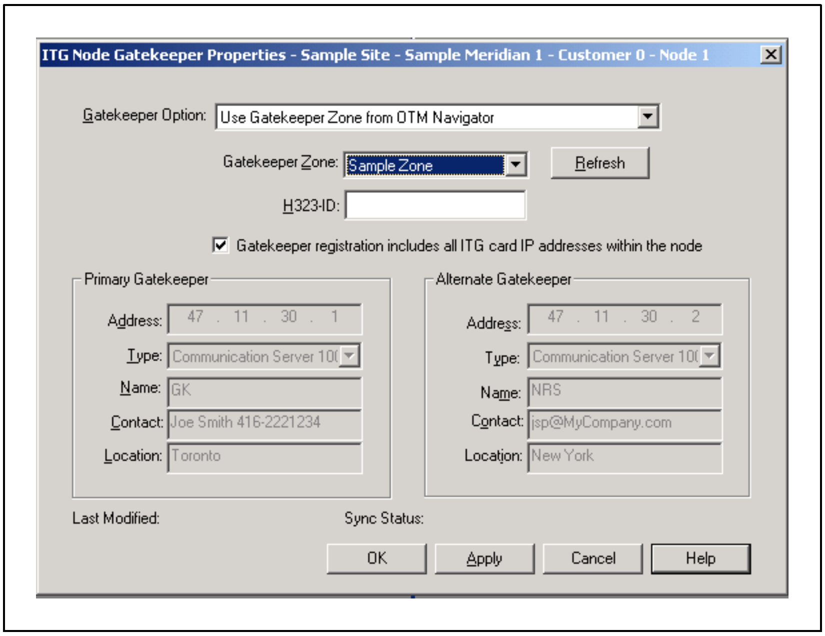
1. The ITG-P 24-port and Media Card 32-port trunk cards must be upgraded to IP Trunk 3.0 (or later) software. This upgrade supports MCDN features and NRS registration. Use TM 2.2 to perform the upgrade. Refer to *Telephony Manager 3.1: System Administration* (NN43050-601) for information on installing, upgrading, and configuring IP Trunk 3.0 (or later) parameters.

2. Configure the IP Trunk 3.0 (or later) node to register with the CS 1000M NRS, using the TM 2.2 ITG Node Gatekeeper Properties window shown in Figure 156 "Gatekeeper Properties window" (page 280). This window enables the administrator to link an IP Trunk 3.0 (or later) endpoint to an NRS (Gatekeeper) zone (automatically providing Primary and Alternate NRSs). This window is also used to manually provision an NRS (Gatekeeper) for the node. Figure 157 "Options in the Gatekeeper Option drop-down list" (page 280) shows the options in the **Gatekeeper Option** drop-down list. Figure 158 "Options in the Primary Gatekeeper's

Type drop-down list" (page 280) shows the options in the **Primary Gatekeeper Type** drop-down list.

Refer to *Telephony Manager 3.1: System Administration* (NN43050-601) for information on how to configure the IP Trunk 3.0 (or later) options.

Figure 156
Gatekeeper Properties window



ITG Node Gatekeeper Properties - Sample Site - Sample Meridian 1 - Customer 0 - Node 1

Gatekeeper Option: Use Gatekeeper Zone from OTM Navigator

Gatekeeper Zone: Sample Zone

H323-ID:

☑ Gatekeeper registration includes all ITG card IP addresses within the node

Primary Gatekeeper

Address: 47 . 11 . 30 . 1

Type: Communication Server 10(

Name: GK

Contact: Joe Smith 416-2221234

Location: Toronto

Alternate Gatekeeper

Address: 47 . 11 . 30 . 2

Type: Communication Server 10(

Name: NRS

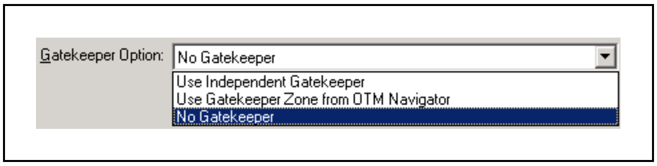
Contact: jsp@MyCompany.com

Location: New York

Last Modified:

Sync Status:

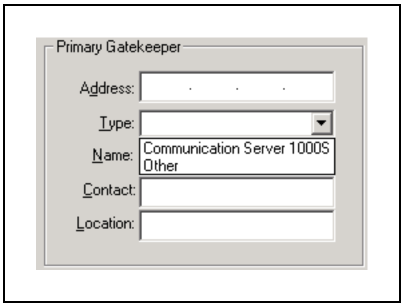
Figure 157
Options in the Gatekeeper Option drop-down list



Gatekeeper Option: No Gatekeeper

Use Independent Gatekeeper
Use Gatekeeper Zone from OTM Navigator
No Gatekeeper

Figure 158
Options in the Primary Gatekeeper Type drop down list



Primary Gatekeeper

Address: . . .

Type:

Communication Server 1000S
Other

Name:

Contact:

Location:

If configured appropriately, the IP Trunk 3.0 (or later) node uses Registration, Admission, and Status signaling (RAS) messaging to register with the NRS. The IP Trunk 3.0 (or later) node then processes calls by scanning its DN information and routing unresolved calls to the NRS, using the Address Translation Protocol Module (ATPM).

The IP Trunk 3.0 (or later) node is subordinate to the NRS for all calls requiring the NRS. The IP Trunk 3.0 (or later) node:

1. registers with the NRS (H.323 Gatekeeper), according to H.323 protocol
2. requests admission
3. accepts the reply, according to H.323 protocol
4. proceeds to handle the call as required, based on the returned message

> ***Note:*** IP Trunk 3.0 (or later) supports the Media Card 32-port trunk card and/or the ITG-P 24-port trunk card.

Refer to *IP Trunk: Description, Installation, and Operation* (NN43001-563) for information on how to install, configure, and operate IP Trunk 3.0 (or later) functions, as well as information on IP Trunk signaling support (for example, MCDN, non-call associated signaling, and ESN5).

Business Communications Manager Release 3.01 (or later)

IP Peer Networking Phase 2 interoperates with BCM Release 3.01 (or later). BCM has been enhanced with many additional MCDN features, including the following:

- Network Call Transfer
- Network Call Redirection Information
- Message Waiting Indication
- ISDN Call Connection Limitation
- Trunk Route Optimization
- Trunk Anti-Tromboning
- Camp-On
- Break-In

For interworking between BCM and a system running CS 1000 Release 4.0 (or later), upgrade the BCM to version 3.01 (or later) software.

A BCM endpoint is configured on the Gatekeeper in the same way that a CS 1000 endpoint is configured. Configure the following on the BCM so that the BCM system can interwork with the CS 1000 Release 4.0 (or later) system:

- Configure **Unified Manager: Services > IP telephony > IP Trunks > H.323 Trunks > Call Signaling** as **GatekeeperRouted** or **GatekeeperResolved**
- Configure **Unified Manager: Services > IP telephony > IP Trunks > H.323 Trunks > Gatekeeper IP** as the IP address of the NRS
- Configure **Unified Manager: Services > IP telephony > IP Trunks > H.323 Trunks > Alias Names** as the Alias name that was used when the H.323 Endpoint for the BCM was created on the NRS

  Note: When working with a BCM 50 system, the Unified Manager is called the BCM Element Manager.

In order to make a BCM 3.01 (or later)-to-CS 1000 call, ensure that the BCM routes and dialing plan (used to reach the CS 1000 systems) match the numbering plan entry assigned to the CS 1000 systems through NRS Manager.

Similarly, to make a CS 1000 system-to-BCM 3.01 call, ensure that the numbering plan entry assigned to the BCM (through NRS Manager) matches the dialing plan information configured on the CS 1000 systems.

Multimedia Communication Server 5100 (MCS 5100)

The SIP Trunk Gateway connects the CS 1000 systems to other Nortel or third-party SIP-enabled products. This direct SIP interface is used to interwork with products such as the MCS 5100. The MCS 5100 brings multimedia features to the CS 1000 system.

For detailed information about MCS 5100 and CS 1000 interworking, refer to *Multimedia Portfolio Communication (MCP) Interworking Basics* (NN42020-127).

Also refer to "Configuring the MCS 5100 system as a Collaborative Server" (page 291).

CallPilot 2.02

CallPilot integrates voice mail, e-mail, and fax messages into a single mailbox. These messages are accessible by telephone, e-mail client, or by any browser-enabled PC.

The SIP Converged Desktop Service (CDS) is a feature convergence of the MCS 5100 and CS 1000 systems. SIP CDS allows users to have simultaneous access to both multimedia features on MCS 5100 and voice features on CS 1000. The CS 1000 system can communicate with the MCS 5100 system by hosting a CallPilot 2.02 mailbox on the CS 1000 system. With SIP, a centralized CallPilot can provide services to a network of CS 1000 and MCS 5100 systems.

CallPilot behind CS 1000

Currently, the unified messaging support for stand-alone MCS 5100 users is provided by a dedicated CallPilot system connected directly to the MCS 5100 system, using a T1/SMDI over IP interface. It is also possible to send Message Waiting Indication (MWI) and call-redirection information to and from a CallPilot behind the CS 1000 system to stand-alone MCS 5100 users, through the SIP interface on the CS 1000 system.

With CallPilot behind CS 1000, all CS 1000 users receive CallPilot service through the existing interface. All MCS 5100 users receive unified messaging services through the SIP Trunk Gateway. For a Converged Desktop user, however, the MWI is sent only to the CS 1000 desktop and is not extended to the SIP client. At this time, the SIP client can get MWI using CallPilot Desktop Manager or My CallPilot (web messaging).

Message Waiting Indication handling

The SIP Trunk Gateway on the CS 1000 system provides MWI service for MCS 5100 remote users served by CallPilot. MCS 5100 users are provisioned on CallPilot, and they are not required to explicitly subscribe MWI service from the CS 1000 SIP Trunk Gateway. When a new message is left for an MCS 5100 user, the CS 1000 system sends an MCDN Facility message with MWI indication to the SIP Trunk Gateway. The message is translated into an unsolicited SIP NOTIFY message with a proper alias address and is sent to the MCS 5100 proxy for further processing. Only the MWI on/off indication is carried in the SIP NOTIFY message.

Subscription for MWI notification is implicit and persistent. The out-of-dialog NOTIFY is used to send MWI notification (that is, the NOTIFY creates its own dialog). The message-summary event package draft defines the structure of the NOTIFY (including its body content). The SIP Trunk Gateway translates the MCDN Facility message to an unsolicited SIP NOTIFY only if the RCAP on D-channel configuration has the MWI settings; otherwise, the SIP Trunk Gateway tunnels the MCDN message into a SIP INVITE message.

Call redirection

MCS 5100 users redirect the call to CallPilot using facilities between the CS 1000 and MCS 5100 systems. The redirecting number is required for the mailbox, and the redirection reason is required for the greeting. The implementation is based on SIP extension headers. In particular, the History header is used to convey the redirection reason (for example, no answer or busy) so that the proper greeting can be played by CallPilot.

CallPilot configuration

Note the following about the CallPilot configuration:

- MCS 5100 users are configured as users on a remote Network Management System (NMS)-node.
- Mailboxes are configured according to the selected numbering plan (UDP or CDP).

For detailed CallPilot configuration information, refer to the following CallPilot NTPs.

- *CallPilot Planning and Engineering Guide (NN44200-200)*
- *CallPilot Installation and Configuration Part 3: T1/SMDI and CallPilot Server Configuration (NN44200-303)*
- *CallPilot Administrator's Guide (44200-601)*

Collaboration between a CS 1000 Release 4.0 (or later) NRS and a Succession 3.0 H.323 Gatekeeper or MCS 5100

A CS 1000 Release 4.0 (or later) NRS is capable of interworking with a Succession 3.0 H.323 Gatekeeper to set up Gatekeeper zones. The Location Request message that is sent between the CS 1000 Release 4.0 (or later) NRS and the Succession 3.0 H.323 Gatekeeper is fully compatible between the two software releases.

> ***Note:*** Collaboration for CDP calls can only be achieved in the same Level 0 Domain (CDP).

As in Succession Release 3.0, there is no need to configure the terminating endpoint on the originating CS 1000 Release 4.0 (or later) NRS (this was only needed for Release 2.0 Gatekeepers).

The following sections provide details for configuring zones between a CS 1000 Release 4.0 (or later) NRS and a Succession Release 3.0 Gatekeeper:

- "Configuring the CS 1000 Release 4.0 (or later) NRS" (page 284)
- "Configuring the Succession Release 3.0 Gatekeeper" (page 290)
- "Configuring the MCS 5100 system as a Collaborative Server" (page 291)

Configuring the CS 1000 Release 4.0 (or later) NRS

To configure a CS 1000 Release 5.0 Linux-based NRS for collaboration with a Succession Release 3.0 Gatekeeper, follow the steps in Procedure 27 "Configuring a Rls 5.0 Linux-based NRS for collaboration with a Rls 3.0 Gatekeeper" (page 285). To configure a CS 1000 Release 4.0 (or later) VxWorks-based NRS for collaboration with a Succession Release 3.0

Gatekeeper, follow the steps in Procedure 28 "Configuring a Rls 4.0 (or later) VxWorks-based NRS for collaboration with a Rls 3.0 Gatekeeper" (page 288).

Procedure 27

Configuring a Rls 5.0 Linux-based NRS for collaboration with a Rls 3.0 Gatekeeper

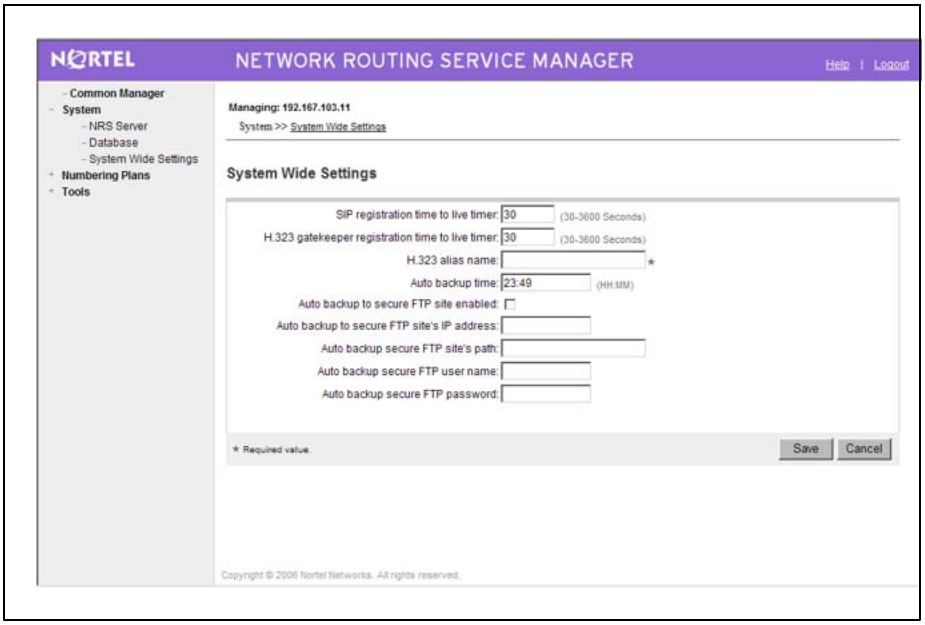
Step	Action

1 Log in to Linux-based NRS Manager. For more information refer to *Network Routing Service Installation and Commissioning (NN43001-564)* .

2 In the NRS Manager Navigator select **System > System Wide Settings**.

The **System Wide Settings** web page opens, as shown in Figure 159 "NRS System Wide Settings web page" (page 285).

Figure 159
NRS System Wide Settings web page



Managing: 192.167.103.11

System >> System Wide Settings

System Wide Settings

SIP registration time to live timer: 30 (30-3600 Seconds)
H.323 gatekeeper registration time to live timer: 30 (30-3600 Seconds)
H.323 alias name: *
Auto backup time: 23:49 (HH:MM)
Auto backup to secure FTP site enabled: □
Auto backup to secure FTP site's IP address:
Auto backup secure FTP site's path:
Auto backup secure FTP user name:
Auto backup secure FTP password:

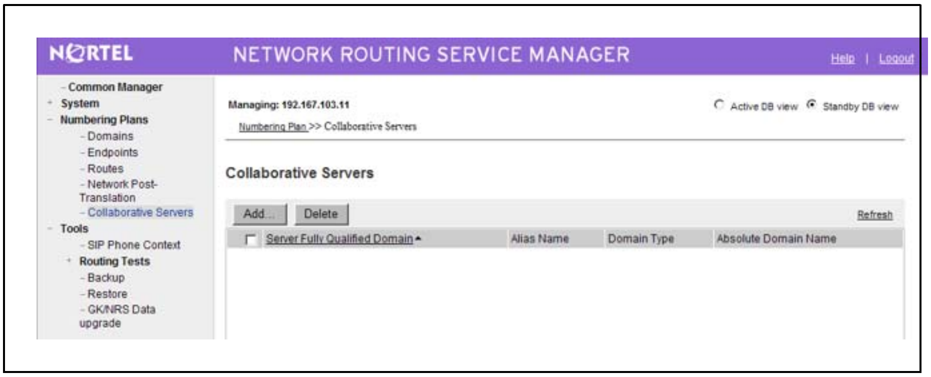
* Required value.

Copyright © 2006 Nortel Networks. All rights reserved.

3 Ensure that the **H.323 alias name** has been entered for the CS 1000 Release 5.0 NRS. The **H.323 alias name** is a mandatory field.

4 In the NRS Manager Navigator select **Numbering Plans > Collaborative Servers**. The Collaborative Servers web page opens, as shown in Figure 160 "Collaborative Servers web page" (page 286).

Figure 160
Collaborative Servers web page



Managing: 192.167.103.11

○ Active DB view ● Standby DB view

Numbering Plan >> Collaborative Servers

Collaborative Servers

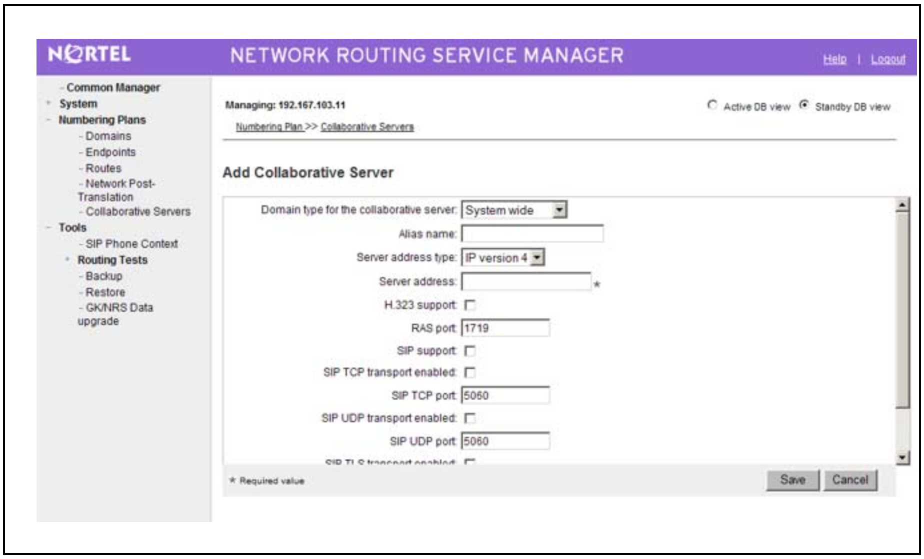
□	Server Fully Qualified Domain ▲	Alias Name	Domain Type	Absolute Domain Name

5 Ensure **Standby database** is selected.

6 Click the **Add....** button.

The **Add Collaborative Server** web page opens, as shown in Figure 161 "Add Collaborative Server web page" (page 286).

Figure 161
Add Collaborative Server web page



Managing: 192.167.103.11

○ Active DB view ● Standby DB view

Numbering Plan >> Collaborative Servers

Add Collaborative Server

Domain type for the collaborative server: System wide
Alias name:
Server address type: IP version 4
Server address: *
H.323 support: □
RAS port: 1719
SIP support: □
SIP TCP transport enabled: □
SIP TCP port: 5060
SIP UDP transport enabled: □
SIP UDP port: 5060
SIP TLS transport enabled: □

* Required value

7 Configure the Succession Release 3.0 Gatekeeper as a collaborative Server. For more information refer to *Network Routing Service Installation and Commissioning (NN43001-564)* .

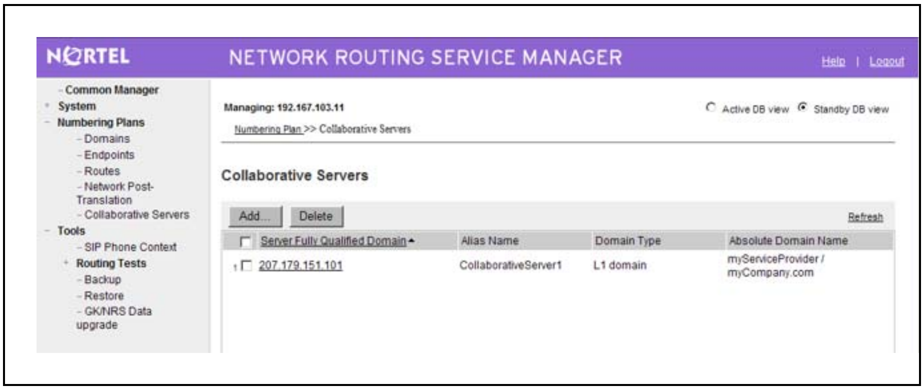
> ***Note 1:*** Ensure that the Succession Release 3.0 Gatekeeper is provisioned in the same Service Domain and Level 1 Domain (UDP) as the originating endpoint.
>
> ***Note 2:*** Provision the Succession Release 3.0 Gatekeeper as a Level 1 Domain Gatekeeper to allow support for CDP interzone

dialing for multiple zones. That is, if the Succession Release 3.0 Gatekeeper supports CDP Domain A and Domain B, then provisioning the Gatekeeper as a Level 1 Zone Collaborative Server allows the CS 1000 Release 5.0 NRS to send it calls from both zones A and B (depending on the CDP domain of the call originator on the CS 1000 Release 5.0 NRS zone).

8 Click **Save**. The standby database is updated.

The **Collaborative Servers** web page opens with the newly added collaborative server, as shown in Figure 162 "Added Collaborative Server web page" (page 287).

Figure 162
Added Collaborative Server web page



Managing: 192.167.103.11

Numbering Plan >> Collaborative Servers

Collaborative Servers

	□ Server Fully Qualified Domain	Alias Name	Domain Type	Absolute Domain Name
1	□ 207.179.151.101	CollaborativeServer1	L1 domain	myServiceProvider / myCompany.com

9 Use a database **Cut over** command to place the database in a Switched Over state. The configuration changes can now be tested. For more information refer to *Network Routing Service Installation and Commissioning (NN43001-564)* .

10 Test the configuration changes.

11 Use a database **Commit** command to update the database with the configuration changes. For more information refer to *Network Routing Service Installation and Commissioning (NN43001-564)* .

—End—

Procedure 28

Configuring a Rls 4.0 (or later) VxWorks-based NRS for collaboration with a Rls 3.0 Gatekeeper

Step	Action

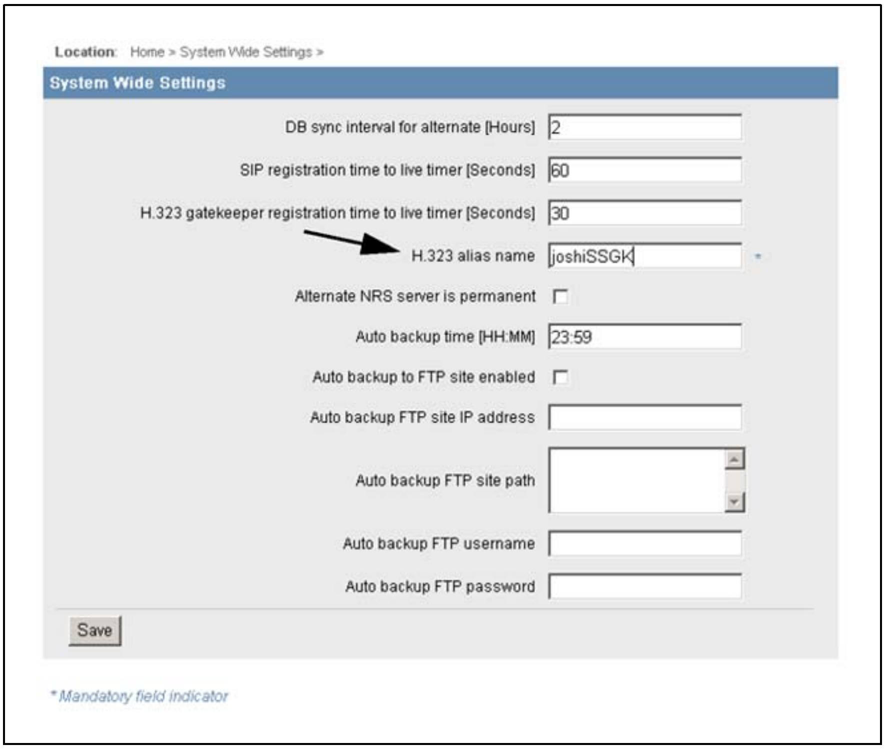
1 Log in to VxWorks-based NRS Manager. For more information refer to *Network Routing Service Installation and Commissioning (NN43001-564)* .

2 Select the **Home** tab on the NRS Manager toolbar.

3 Select **System Wide Settings** in the NRS Manager Navigator. For more information refer to *Network Routing Service Installation and Commissioning (NN43001-564)* .

4 Ensure that the **H.323 alias name** has been entered for the CS 1000 Release 4.0 (or later) NRS (see Figure 163 "NRS System Wide Settings web page" (page 288)).

Figure 163
NRS System Wide Settings web page



Location: Home > System Wide Settings >

System Wide Settings

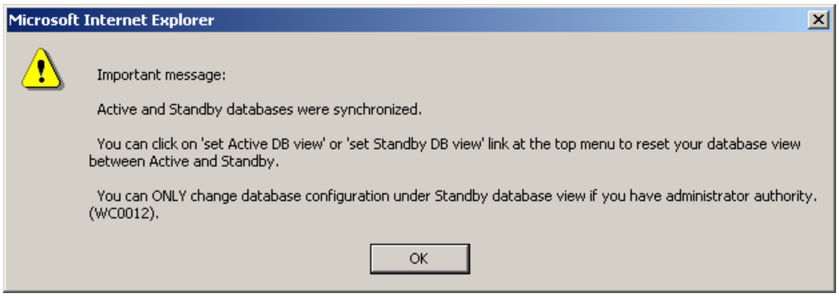
DB sync interval for alternate [Hours]	2
SIP registration time to live timer [Seconds]	60
H.323 gatekeeper registration time to live timer [Seconds]	30
H.323 alias name	joshiSSGK *
Alternate NRS server is permanent	□
Auto backup time [HH:MM]	23:59
Auto backup to FTP site enabled	□
Auto backup FTP site IP address	
Auto backup FTP site path	
Auto backup FTP username	
Auto backup FTP password	

* Mandatory field indicator

5 Select the **Configuration** tab on the NRS Manager toolbar. Click **OK**.

A dialog box opens indicating the status of the active and standby database, as shown in Figure 164 "Configuration tab message" (page 289).

Figure 164
Configuration tab message



Important message:

Active and Standby databases were synchronized.

You can click on 'set Active DB view' or 'set Standby DB view' link at the top menu to reset your database view between Active and Standby.

You can ONLY change database configuration under Standby database view if you have administrator authority. (WC0012).

6 Click **OK**.

7 Switch to **Standby DB view**.

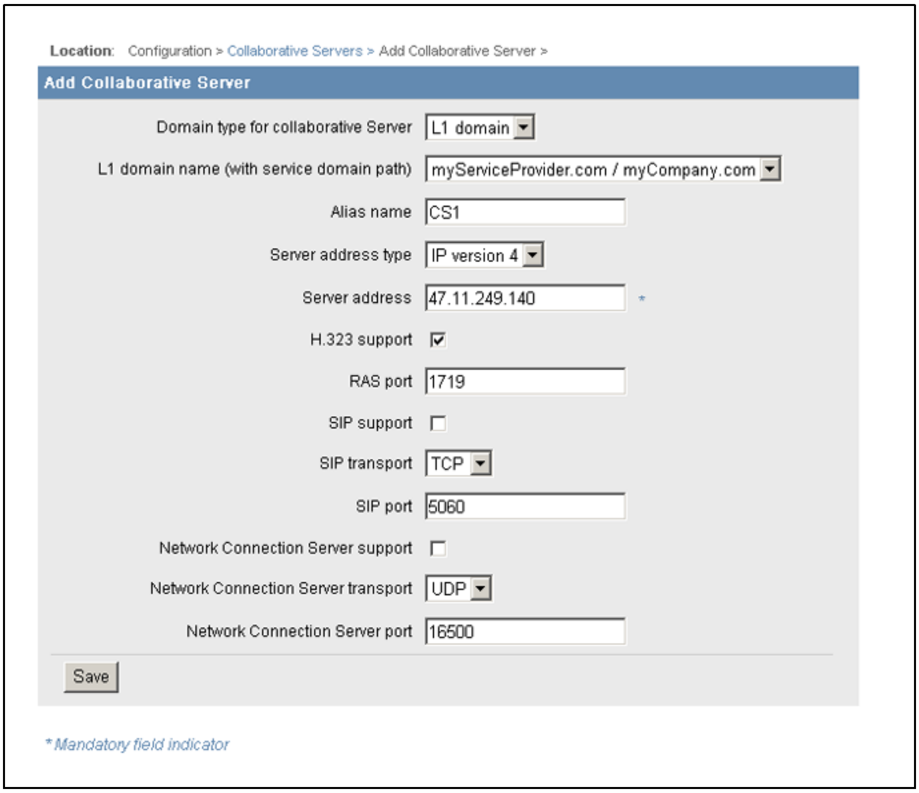
8 Select **Collaborative Servers** in the NRS Manager Navigator.

9 Click **Add**. The **Add Collaborative Servers** web page opens, as shown in Figure 165 "Add Collaborative Servers web page" (page 290).

10 Configure the Succession Release 3.0 Gatekeeper as a collaborative Server. For more information refer to *Network Routing Service Installation and Commissioning (NN43001-564)* .

> ***Note 1:*** Ensure that the Succession Release 3.0 Gatekeeper is provisioned in the same Service Domain and Level 1 Domain (UDP) as the originating endpoint.
>
> ***Note 2:*** Provision the Succession Release 3.0 Gatekeeper as a Level 1 Domain Gatekeeper to allow support for CDP interzone dialing for multiple zones. That is, if the Succession Release 3.0 Gatekeeper supports CDP Domain A and Domain B, then provisioning the Gatekeeper as a Level 1 Zone Collaborative Server allows the CS 1000 Release 4.0 (or later) NRS to send it calls from both zones A and B (depending on the CDP domain of the call originator on the CS 1000 Release 4.0 [or later] NRS zone).

Figure 165
Add Collaborative Servers web page



Location: Configuration > Collaborative Servers > Add Collaborative Server >

Add Collaborative Server

Domain type for collaborative Server	L1 domain
L1 domain name (with service domain path)	myServiceProvider.com / myCompany.com
Alias name	CS1
Server address type	IP version 4
Server address	47.11.249.140 *
H.323 support	☑
RAS port	1719
SIP support	□
SIP transport	TCP
SIP port	5060
Network Connection Server support	□
Network Connection Server transport	UDP
Network Connection Server port	16500

* Mandatory field indicator

11 Select the **Tools** tab on the NRS Manager toolbar.

12 Select **Database Actions** in the NRS Manager Navigator.

13 Use the `Commit` command if you are sure the configuration is correct. Otherwise, first use the `Cutover` command and test the configuration changes.

—End—

Configuring the Succession Release 3.0 Gatekeeper

Procedure 29
Configuring the Succession Release 3.0 Gatekeeper

Step Action

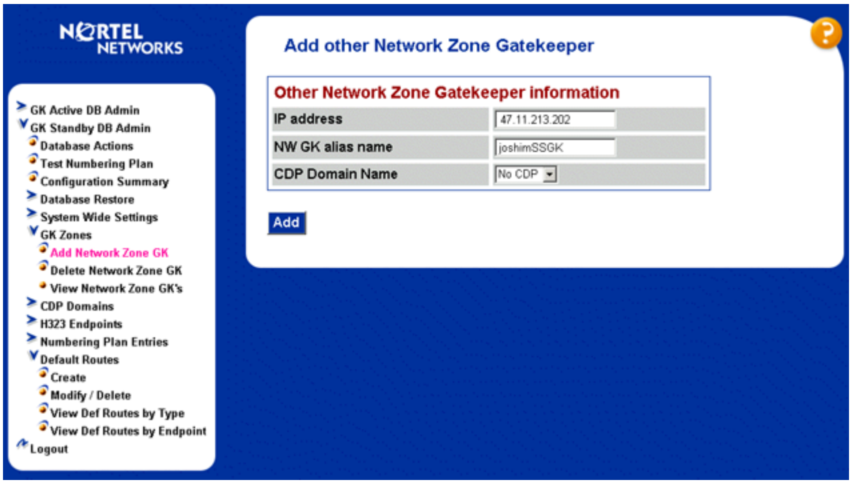
1 Log in to the **Gatekeeper** web pages in Element Manager (for Succession Release 3.0).

2 Select the **GK Standby DB Admin > GK Zones > Add Network Zone GK** link (see Figure 166 "Add other Network Zone Gatekeeper" (page 291)).

3 Add the CS 1000 Release 4.0 (or later) NRS as a Network Zone Gatekeeper using the same Gatekeeper H.323 alias name as configured in step 4.

> ***Note:*** A Succession Release 3.0 Network Zone Gatekeeper is similar to a CS 1000 Release 4.0 (or later) Collaborative Server.

Figure 166
Add other Network Zone Gatekeeper



Add other Network Zone Gatekeeper

Other Network Zone Gatekeeper information	
IP address	47.11.213.202
NW GK alias name	joshimSSGK
CDP Domain Name	No CDP

4 Select the **GK Standby Database Admin > Database Actions** link.

5 Click the **SingleStepCutoverCommit** command once you are sure that the configuration is correct. Otherwise, first use the `Cutover` command to test the configuration.

—End—

Configuring the MCS 5100 system as a Collaborative Server

The MCS 5100 can be configured as a collaborative server which supports H.323 and SIP.

The configuration for H.323 is the same as configuring another Succession Release 3.0 Gatekeeper with the limitation that MCS 5100 can only be configured as a Level 0 Domain Collaborative Server. The MCS 5100 cannot parse the H.323 LRQ messages which specify a Level 0 Domain on a per-call basis. So it always routes the call to a fixed Level 0 Domain. So in this case, it is only the MCS 5100 H.323 alias name (which is configured as the Collaborative Server Gatekeeper alias name) and IP address that is relevant.

For SIP calls, the MCS 5100 can be configured as a Level 1 or Level 0 Domain Collaborative Server. The Service Domain on the NRS should match the Service Domain on the MCS 5100. A Level 1 Domain is preferred, so the MCS 5100 can be used for calls originating from multiple Level 0 Domains without restriction.

Maintenance

Contents

This section contains information on the following topics:

Command Line Interface commands

The Signaling Server provides a Command Line Interface (CLI) through a serial port or a Telnet session. This section contains the VxWorks CLI commands available at that interface that are applicable to IP Peer Networking.

Signaling Server VxWorks CLI commands are available at three levels:

- Level One — Operations, Administration, and Maintenance (OAM) shell for basic technician support and general status system checking (**`oam>prompt`**)
- Level Two — Problem Determination Tool (PDT) shell for expert support; also includes all Level One (OAM) commands (**`pdt>prompt`**)
- Level Three — Nortel proprietary vxWorksTa shell for advanced debugging and design support (**`prompt`**)

  Note: This section describes the Level One (OAM) and Level Two (PDT) CLI commands. Level Three commands are considered expert support and design level commands, and are not documented here.

You must log in to the Signaling Server to use the VxWorks CLI commands. Refer to *Signaling Server Installation and Commissioning* (NN43001-312) for this procedure.

Help CLI commands

Table 13 "Help CLI commands" (page 294) includes the general help CLI commands. These commands are available only at the OAM and PDT shells.

Table 13
Help CLI commands

CLI Command	Description
help	Lists all command groups available at current shell level.
help <command>	Provides Help text on a particular command.

Virtual Trunk CLI commands

Table 14 "Virtual Trunk CLI commands" (page 294) includes the CLI commands used when working with Virtual Trunks.

Table 14
Virtual Trunk CLI commands

CLI Command	Description
help vtrk	Lists all Virtual Trunk-related commands (for example, vtrkShow). This help command is available at the OAM and PDT shells.

CLI Command	Description
	Note: The Virtual Trunk group includes both H.323 and SIP Virtual Trunk commands.
vtrkShow <protocol>, <start#>, <howMany>	Provides a summary of the Virtual Trunk configuration of a particular protocol. Where: • protocol is either SIP or H.323. If the protocol parameter is omitted, then the command prints a summary of both the H.323 and SIP trunks. Otherwise, the command prints the specified protocol. • start# specifies that the printing starts from specified channel ID. If the start# parameter is omitted, then the command starts from the first channel of specific protocol. • howMany specifies the number of channels to be printed. If the howMany parameter is omitted, then the command prints all channels for specified protocol starting from the start#.

D-channel CLI command

Table 15 "D-channel CLI commands" (page 295) includes D-channel CLI commands

Table 15
D-channel CLI commands

CLI Command	Description
DCHmenu	Displays a menu to perform various information retrieval operations for the D-channel. The output for DCHmenu: oam->DCHmenu Please select one of the DCHmenu options: 0 - Print menu (default) 1 - Print current DCH state 2 - Print current DCH configuration 3 - Print application error log

CLI Command	Description
	4 - Print link error log 5 - Print protocol error log 6 - Print message log 7 - Enable printing all messages processed by UIPC 8 - Enable error printing 9 - Enable info printing 10 - Enter manual message mode 11 - Print b channel control blocks 99 - Exit menu Please enter your DCHmenu choice (0 to print the menu): 1

H323GwShow CLI commands

Table 16 "H323GwShow trace tool CLI commands" (page 296) includes the H323GwShow trace tool CLI commands applicable to the Signaling Server. The commands are issued from the OAM shell.

Table 16
H323GwShow trace tool CLI commands

CLI Command	Description
H323GwShow	Provides a general summary of the H.323 Virtual Trunk settings.
H323GwShow ch <channel #>	Provides a snapshot summary of the state of the H.323 Virtual Trunk setting and a snapshot of the active call on the specified channel (if the call exists). Where channel # indicates the channel number to trace. The values range from 0 - maximum channel number.
H323GwShow num <calling/called number>	Provides a snapshot summary of the state of the H.323 Virtual Trunk settings and a snapshot of the active calls using the calling/called number or partial number specified. Where calling/called number indicates the telephone number to trace. The value can be a number from 1 to 32 digits and can be a partial calling/called number.

CLI Command	Description
H323GwShow num <calling/called number> <NPI> <TON>	Provides a snapshot summary of the state of the H.323 Virtual Trunk settings. It also provides a snapshot of the active calls using the calling/called number or partial number with the specified NPI and TON values. Where: • calling/called number indicates the telephone number to trace. The value can be a number from 1 to 32 digits and can be a partial calling/called number. • NPI specifies the numbering plan identifier for the calls. The calls using this numbering plan are to be traced. The values are: 0 - ALL 1 - Unknown 2 - ISDN/telephony numbering plan (E.164) 4 - E.163 5 - Telex numbering plan (F.69) 6 - Data numbering plan 7 - National standard numbering plan • TON specifies the type of number to use as a filter for tracing. Only calls using this TON setting are traced. The values are: 0 - ALL 1 - Unknown Number 2 - International Number 3 - National Number 4 - Network Specific Number 5 - Subscriber Number 6 - Level 1 Regional 7 - Level 0 Interface
help H323GwShow	Displays the usage of the H323GwShow commands.

SIPGwShow CLI commands

Table 16 "H323GwShow trace tool CLI commands" (page 296) includes the SIPGwShow trace tool CLI commands applicable to the Signaling Server. The commands are issued from the OAM shell.

Table 17
SIPGwShow trace tool CLI commands

CLI Command	Description
SIPGwShow	Shows the general SIP Virtual Trunk settings.

CLI Command	Description
SIPGwShow ch <channel #>	Provides a snapshot summary of the SIP Virtual Trunk configuration for the specific channel ID. The command also provides a snapshot of the active call on the specified channel (if the call exists). The channel # indicates the channel number to trace. The values range from 0 - maximum channel number.
SIPGwShow num <calling/called number>	Provides a snapshot summary of the SIP Virtual Trunk configuration for the specific calling-party or called-party number. The command also provides a snapshot of the active calls using the calling-party/called-party number or partial number specified. The calling/called number indicates the telephone number to trace. The value can be a number from 1 to 32 digits and can be a partial calling/called number.
SIPGwShow num <calling/called number> <NPI> <TON>	Provides a snapshot summary of the SIP Virtual Trunk configuration. The command also provides a snapshot of the active calls using the calling/called number or partial number with the specified NPI and TON values. Where: • calling/called number indicates the telephone number to trace. The value can be a number from 1 to 32 digits and can be a partial calling/called number. • NPI specifies the numbering plan identifier for the calls to be traced. The values are: 0 - ALL 1 - Unknown 2 - ISDN/telephony numbering plan (E.164) 4 - E.163 5 - Telex numbering plan (F.69) 6 - Data numbering plan 7 - National standard numbering plan • TON specifies the type of number to use as a filter for tracing. Only calls using this TON setting are traced. The values are: 0 - ALL 1 - Unknown Number 2 - International Number 3 - National Number 4 - Network Specific Number 5 - Subscriber Number 6 - Level 1 Regional 7 - Level 0 Interface
help SIPGwShow	Displays the usage of the SIPGwShow commands.

Graceful disable CLI commands

Table 18 "Graceful Disable commands" (page 299) includes graceful disable CLI commands applicable to the Signaling Server. They are issued from the OAM shell.

Table 18
Graceful Disable commands

CLI Command	Description
disServices	Causes the server to gracefully switch all registered resources (including telephone, Virtual Trunk, and Voice Gateways) to the other services (Signaling Server or Voice Gateway Media Card) in the same node. This command should not interrupt existing calls.
disTPS	Causes the line TPS to gracefully switch the registered telephones to the other cards located in the same node. On the Signaling Server the TPS disable command affects only the LTPS. It does not affect the virtual trunks or gatekeeper components, which means the node mastership is not moved to another LTPS. TPS disable command is only intended to disable phone registration on current device. It is not intended to call election, unless no VTRK is configured on this device.
disVTRK	Causes the Virtual Trunk to gracefully switch the registered Virtual Trunks to other Signaling Servers located in the same node. ***Note:*** LTPS and VTRK functions must be enabled on a Signaling Server located in the same node to accept VTRK registrations. The number of VTRK resources available must be equal to or greater than the number of VTRK resources being switched over.
forcedisServices	Forces the server to switch all registered resources to another Signaling Server or Voice Gateway Media Card in the same node. This command causes any existing calls to be dropped.
forcedisTPS	Forces all registered line LTPS to unregister from the local server. On the Signaling Server the TPS disable command affects only the LTPS. It does not affect the virtual trunks or gatekeeper components, which means the node mastership is not moved to another LTPS. TPS disable command is only intended to disable phone registration on current device. It is not intended to call election, unless no VTRK is configured on this device.
forcedisVTRK	Forces all registered Virtual Trunks to unregister from the local server.
enlServices	Causes all the services to accept registration of resources.

CLI Command	Description
enlTPS	Causes line TPS application to be enabled and to accept set registrations.
enlVTRK	Causes the Virtual Trunk application to be enabled and to accept Virtual Trunk registrations.
loadBalance	Causes the service to attempt to balance the registration load of sets between this service and the rest of the node services.
servicesStatusShow	Shows the status of services (tps/iset/vtrk/gk)
soHelpMenu	Displays all the commands that can be used for Services Switch-Over.

Trace tools CLI commands

The following section outlines the CLI commands for the message trace tools.

- Table 19 "General trace tool CLI commands" (page 300) shows the general trace tool commands.
- Table 20 "Gatekeeper protocol trace tool CLI commands" (page 301) shows the Gatekeeper protocol trace tool commands.
- Table 21 "SIPCallTrace trace tool CLI commands" (page 305) shows the SIP trace tool commands.
- Table 22 "H.323CallTrace trace tool CLI commands" (page 310) shows the H.323 trace tool commands.
- Table 23 "NCS CLI commands" (page 312) shows the Network Connection Service (NCS) trace tool commands

  Note: A warm boot of the system causes all tracing to cease. Traces must be entered again after the system has restarted.

General trace tool commands

Table 19 "General trace tool CLI commands" (page 300) includes the general trace tool CLI commands applicable to the Signaling Server and the Voice Gateway Media Cards. The commands are issued from the OAM shell of the Signaling Server and the LTPS prompt of the Voice Gateway Media Cards.

Table 19
General trace tool CLI commands

CLI Command	Description
traceShow	Displays the names of active traces in the system.
traceAllOff	Causes all traces that use the monitorLib server to stop their output.

CLI Command	Description
tracePrintOff	Blocks all logging of information received by the monitorLib service to the TTY output. This does not include traces directed through the monitorLib service to the RPT.LOG or SYSLOG.n services.
traceFileOff	Causes the monitorLib server to stop logging to the log files any and all trace information received by the service. The log files include syslog.n for the Voice Gateway Media Card and rpt.log for the Signaling Server.
traceAllOn	Clears the blocking of all trace information imposed on the monitorLib service by the traceAllOff command, the tracePrintOff command, and the traceFileOff command. By default, all tracing is on.
tracePrintOn	Clears only the TTY output blocking that was imposed by the traceAllOff and tracePrintOff commands.
traceFileOn	Clears only the blocking of logging to files that was imposed by the traceAllOff and traceFileOff commands.

Note 1: A warm boot of the system causes all tracing to cease. Traces must be entered again after the system has restarted.

Note 2: If no directory path is supplied with the filename specified, then the file is written to the C:/U/trace directory on the Voice Gateway Media Cards and to the /u/trace directory on the Signaling Server.

Note 3: If no filename is given, then no trace file is generated and output is directed to the TTY. If the filename does not meet the DOS 8.3 restriction, then the filename is rejected and no file is generated. If the file is deleted, cannot be found, or has a write error, then the output is directed to the TTY.

Note 4: If the output for the trace cannot be determined, then the output is directed to the TTY.

Gatekeeper protocol trace tool commands

Table 20 "Gatekeeper protocol trace tool CLI commands" (page 301) includes the protocol trace tool CLI commands for the Gatekeeper. These commands are issued from the OAM shell.

Table 20
Gatekeeper protocol trace tool CLI commands

CLI Command	Description
gkDiscoveryTrace ID <"Alias Name"> IP <"IP address">	The trace outputs the GRQ, GCF, and GRJ messages for the specified endpoint. Where: • Alias Name is the H.323 ID string. • IP address is the endpoint’s IP address.

CLI Command	Description
ALL	• ALL causes a trace on all endpoints.
gkRegTrace ID <"Alias Name"> IP <"IP address"> ALL	The trace outputs the RRQ, RCF, RRJ, URQ, UCF, and URJ messages for the specified endpoint. Where: • Alias Name is the H.323 ID string. • IP address is the endpoint's IP address. • ALL causes a trace on all endpoints.
gkCallTrace ID <"Alias Name"> IP <"IP address"> NUM <calling/called Number> NUM <calling/called Number> <NPI> <TON> ALL	The trace outputs the ARQ, ACF, ARJ, LRQ, LCF, LRJ, BRQ, BCF, BRJ, DRQ, DCF, and DRJ messages for the specified endpoint. Where: • Alias Name is the H.323 ID string. • IP address specifies the endpoint's IP address. • calling/called number indicates the telephone number to trace. The value can be a number from 1 to 32 digits and can be a partial calling/called number. • NPI - Specifies the numbering plan identifier for which calls using this numbering plan are to be traced. The values are: 0 - ALL 1 - Unknown 2 - ISDN/telephony numbering plan (E.164) 4 - E.163 5 - Telex numbering plan (F.69) 6 - Data numbering plan 7 - National standard numbering plan • TON - specifies the type of number to use as a filter for tracing. Only calls using this TON setting are traced. The values are: 0 - ALL 1 - Unknown Number 2 - International Number 3 - National Number 4 - Network Specific Number 5 - Subscriber Number 6 - Level 1 Regional 7 - Level 0 Interface • ALL - Causes a trace on all endpoints ***Note:*** A maximum of ten number traces are allowed.

CLI Command	Description
gkProtocolTrace ID <"Alias Name"> <"protocol"> IP <"IP address"> <"protocol"> NUM <calling/called Number> <"protocol"> NUM <calling/called Number> <NPI> <TON> <"protocol"> ALL <protocol>	Traces messages for the specified endpoint. Where: • Alias Name is the H.323 ID string. • IP address specifies the endpoint's IP address. • calling/called number indicates the telephone number to trace. The value can be a number from 1 to 32 digits and can be a partial calling/called number. • NPI specifies the numbering plan identifier for which calls using this numbering plan are to be traced. The values are: 0 - ALL 1 - Unknown 2 - ISDN/telephony numbering plan (E.164) 4 - E.163 5 - Telex numbering plan (F.69) 6 - Data numbering plan 7 - National standard numbering plan
	• TON - specifies the type of number to use as a filter for tracing. Only calls using this TON setting are traced. The values are: 0 - ALL 1 - Unknown Number 2 - International Number 3 - National Number 4 - Network Specific Number 5 - Subscriber Number 6 - Level 1 Regional 7 - Level 0 Interface • ALL causes a trace on all endpoints • protocol - specifies which protocols to trace. (1) Valid protocol types for IP and ALL tracing — individually: ALL, ARQ, ACF, ARJ, BRQ, BCF, BRJ, DRQ, DCF, DRJ, GRQ, GCF, GRJ, LRQ, LCF, LRJ, NSM, RRQ, RCF, RRJ, RIP, URQ, UCF, AND URJ are acceptable inputs. — by category: AXX – ARQ, ACF, ARJ BXX – BRQ, BCF, BRJ DXX – DRQ, DCF, DRJ GXX – GRQ, GCF, GRJ LXX – LRQ, LCF, LRJ RXX – RRQ, RCF, RRJ UXX – URQ, UCF, URJ

CLI Command	Description
	(2) Valid protocols for NUM tracing — Individually: ALL, ARQ, ACF, ARJ, BRQ, BCF, BRJ, DRQ, DCF, DRJ,LRQ, LCF, LRJ — by category: AXX – ARQ, ACF, ARJ BXX – BRQ, BCF, BRJ DXX – DRQ, DCF, DRJ LXX – LRQ, LCF, LRJ (3) Protocols that cannot be traced by any endpoint: — IACK, INAC, IRQ, IRR, RAI, RAC, SCI, SCR, XRS To trace multiple protocols, separate the input with a space (for example, ARQ ACF ARJ). ***Note:*** A maximum of ten number traces are allowed.
gkTraceOff IP <"IP address"> ALL	Turns off the trace for the specified endpoint.
gkTraceOutput <Output_Destination> <"File Pathname">	Sets the output for all gk protocol traces. Where: • Output_Destination specifies where all the trace messages for the gkTrace are to be directed. Values are: 1 = TTY 2 = RPTLOG 3 = File 4 = File and TTY • "File Pathname" is a string encapsulated in quotes. It specifies the file to output to if option 3 or 4 was selected
gkTraceSettings	Displays all endpoints that are being traced. The command also displays the location where the output is sent (TTY, RPT.LOG, or a file and the file’s location).

CLI Command	Description
gkTraceTblClear	Clears the calling/called number table associated with the NUM trace filter(s). A maximum of 200 tables entries are allowed. If there are more than 200 table entries, the system displays the following error: gkTrace callIdentifier table is full Clearing the table is a temporary solution. Better options may include: • refining the NUM trace filter to be more exact • reducing the number of NUM trace filters • running the trace during lower traffic periods
gkTraceTblShow	Displays the calling/called number table associated with the NUM trace filter(s). Some entries may be shown twice, since intrazone calls generate two ARQ messages. Interzone calls generate only one ARQ message.

Note: A warm boot of the system causes all tracing to cease. Traces must be entered again after the system restarts.

SIPCallTrace trace tool commands

Table 21 "SIPCallTrace trace tool CLI commands" (page 305) includes the SIPCallTrace trace tool CLI commands applicable to the Signaling Server. They are issued from the OAM shell.

Table 21
SIPCallTrace trace tool CLI commands

CLI Command	Description
SIPCallTrace on	Turns on SIP Virtual Trunk tracing for all channels.
SIPCallTrace off	Turns off SIP Virtual Trunk tracing for all channels.
SIPTraceLevel <Output Option>	Sets the SIPCallTrace output to Summary or Detailed format. The Summary format provides only information normally displayed by the SIPCallTrace command. The Detailed format provides a more detailed output of the SIP signaling messages associated with the traces that are set using the SIPCallTrace utility. Output Option specifies the level of the SIP Message Trace. The values are: 0 = SIP Message Trace – Summary (default)

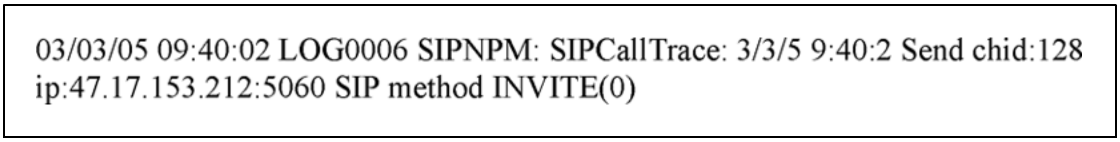
CLI Command	Description
	1 = SIP Message Trace – Detailed See Figure 167 "SIPCallTrace output example (INVITE message only) — Summary format" (page 309) for an example of SIPCallTrace in Summary format. See Figure 168 "SIPCallTrace output example (INVITE message only) — Detailed format" (page 309) for an example of SIPCallTrace in Detailed format.
help SIPTraceLevel	Displays the usage for the SIPTraceLevel CLI command.
SIPCallTrace <MsgRecv> <MsgSend>	Allows tracing of all SIP channels in the receiving and/or sending directions. Where: • MsgRecv specifies if the messages sent to the specified channel should be traced. The values are ON or OFF. • MsgSend specifies if the messages sent from the specified channel should be traced. The values are ON or OFF.
SIPCallTrace ch <channel #> <MsgRecv> <MsgSend>	Allows tracing of a specified SIP channel in the receiving and/or sending directions. Where: • channel # indicates the channel number to trace. The values range from 0 - maximum channel number. • MsgRecv specifies if the messages sent to the specified channel should be traced. The values are ON or OFF. • MsgSend specifies if the messages sent from the specified channel should be traced. The values are ON or OFF.
SIPCallTrace num <calling/called number> <MsgRecv> <MsgSend>	Allows the tracing of SIP messages using the called and calling numbers in the receiving and/or sending directions. If the called or calling number of a SIP Virtual Trunk session matches the number specified, then the messages to and from the Virtual Trunk are traced. Where: • calling/called number indicates the telephone number to trace on. The number can be from 1 to 32 numeric digits and can be a partial calling/called number.

CLI Command	Description
	• MsgRecv specifies if the messages sent to the specified channel should be traced. The values are ON or OFF. • MsgSend specifies if the messages sent from the specified channel should be traced. The values are ON or OFF.
SIPCallTrace ch <beginning channel #> <ending channel #> <MsgRecv> <MsgSend>	Allows the tracing of a range of SIP Virtual Trunk channels in the receiving and/or sending directions. Where: • beginning channel # indicates the channel number to trace. The values range from 0 - maximum channel number. • ending channel # indicates the channel number to trace. The values range from 0 - maximum channel number, but must be greater than the beginning channel #. • MsgRecv specifies if the messages sent to the specified channel should be traced. The values are ON or OFF. • MsgSend specifies if the messages sent from the specified channel should be traced. The values are ON or OFF.
SIPCallTrace num <calling/called number> <NPI> <TON> <MsgRecv > <MsgSend>	Allows a user to trace SIP messages using the called and calling numbers in the receiving and/or sending directions. If the called or calling number of a SIP Virtual Trunk session matches the number specified and the specified NPI and TON values match the call type, then the messages to and from the SIP Virtual Trunk are traced. Where: • calling/called number indicates the telephone number to trace on. The number can be from 1 to 32 numeric digits and can be a partial calling/called number. • NPI specifies the numbering plan identifier for which calls using this numbering plan are to be traced. The values are: 0 - ALL 1 - Unknown 2 - ISDN/telephony numbering plan (E.164) 4 - E.163 5 - Telex numbering plan (F.69) 6 - Data numbering plan 7 - National standard numbering plan • TON - specifies the type of number to use as a filter for tracing. Only calls using this TON setting are traced. The values are:

CLI Command	Description
	0 - ALL 1 - Unknown Number 2 - International Number 3 - National Number 4 - Network Specific Number 5 - Subscriber Number 6 - Level 1 Regional 7 - Level 0 Interface • MsgRecv specifies if the messages sent to the specified channel should be traced. The values are ON or OFF. • MsgSend specifies if the messages sent from the specified channel should be traced. The values are ON or OFF.
help SIPCallTrace	Provides a description of the SIPCallTrace commands. It also supplies parameters and possible parameter values for each command. The command also supplies a list of associated CLIs (that is, SIPTraceShow and SIPOutput).
SIPOutput <Output_Destination> <"File Pathname">	Specifies where the output for the trace tool is to be directed. Where: • Output_Destination specifies where all the trace messages for the SIPCallTrace are to be directed. The values are: 1 = TTY 2 = RPTLOG 3 = File 4 = File+TTY • "File Pathname" is a string encapsulated in quotes. The file pathname must be specified if option 3 (File) was selected as the output destination.
help SIPOutput	Displays the usage for the SIPOutput CLI command.
SIPTraceShow	Displays the SIP trace settings, including the output format, output destination and filename, as well as all active traces for the SIPCallTrace trace tool.
help SIPTraceShow	Displays the usage for the SIPTraceShow CLI command.

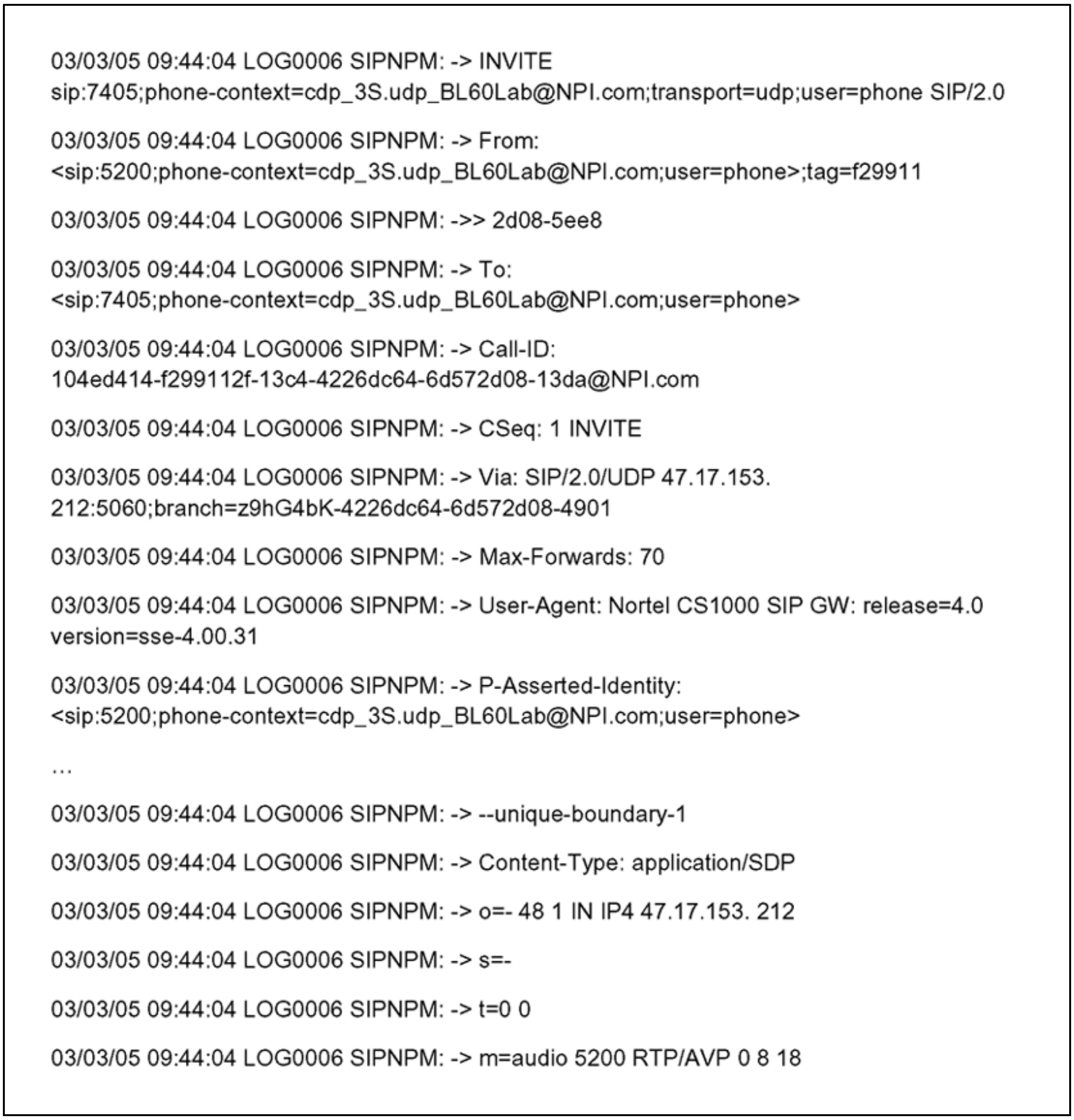
Figure 167 "SIPCallTrace output example (INVITE message only) — Summary format" (page 309) shows an example of SIPCallTrace results in Summary format. Figure 168 "SIPCallTrace output example (INVITE message only) — Detailed format" (page 309) shows an example of SIPCallTrace results in Detailed format.

Figure 167
SIPCallTrace output example (INVITE message only) Summary format



```
03/03/05 09:40:02 LOG0006 SIPNPM: SIPCallTrace: 3/3/5 9:40:2 Send chid:128
ip:47.17.153.212:5060 SIP method INVITE(0)
```

Figure 168
SIPCallTrace output example (INVITE message only) Detailed format



```
03/03/05 09:44:04 LOG0006 SIPNPM: -> INVITE
sip:7405;phone-context=cdp_3S.udp_BL60Lab@NPI.com;transport=udp;user=phone SIP/2.0

03/03/05 09:44:04 LOG0006 SIPNPM: -> From:
<sip:5200;phone-context=cdp_3S.udp_BL60Lab@NPI.com;user=phone>;tag=f29911

03/03/05 09:44:04 LOG0006 SIPNPM: ->> 2d08-5ee8

03/03/05 09:44:04 LOG0006 SIPNPM: -> To:
<sip:7405;phone-context=cdp_3S.udp_BL60Lab@NPI.com;user=phone>

03/03/05 09:44:04 LOG0006 SIPNPM: -> Call-ID:
104ed414-f299112f-13c4-4226dc64-6d572d08-13da@NPI.com

03/03/05 09:44:04 LOG0006 SIPNPM: -> CSeq: 1 INVITE

03/03/05 09:44:04 LOG0006 SIPNPM: -> Via: SIP/2.0/UDP 47.17.153.
212:5060;branch=z9hG4bK-4226dc64-6d572d08-4901

03/03/05 09:44:04 LOG0006 SIPNPM: -> Max-Forwards: 70

03/03/05 09:44:04 LOG0006 SIPNPM: -> User-Agent: Nortel CS1000 SIP GW: release=4.0
version=sse-4.00.31

03/03/05 09:44:04 LOG0006 SIPNPM: -> P-Asserted-Identity:
<sip:5200;phone-context=cdp_3S.udp_BL60Lab@NPI.com;user=phone>

...

03/03/05 09:44:04 LOG0006 SIPNPM: -> --unique-boundary-1

03/03/05 09:44:04 LOG0006 SIPNPM: -> Content-Type: application/SDP

03/03/05 09:44:04 LOG0006 SIPNPM: -> o=- 48 1 IN IP4 47.17.153. 212

03/03/05 09:44:04 LOG0006 SIPNPM: -> s=-

03/03/05 09:44:04 LOG0006 SIPNPM: -> t=0 0

03/03/05 09:44:04 LOG0006 SIPNPM: -> m=audio 5200 RTP/AVP 0 8 18
```

H.323CallTrace trace tool commands

Table 22 "H.323CallTrace trace tool CLI commands" (page 310) includes the H.323 trace tool CLI commands applicable to the Signaling Server. They are issued from the OAM shell.

Table 22
H.323CallTrace trace tool CLI commands

CLI Command	Description
H323CallTrace ch <channel #> <MsgRecv> <MsgSend>	Traces a specified channel. Where: • channel # indicates the channel number to trace. Values range from 0 - maximum channel number. • MsgRecv specifies if the messages sent to the specified channel should be traced. The values are ON or OFF. • MsgSend specifies if the messages sent from the specified channel should be traced. The values are ON or OFF. ***Note:*** Replaces the H323CallTrace <channel #> <MsgRecv > <MsgSend> command.
H323CallTrace num <calling/called number> <MsgRecv> <MsgSend>	Traces H.323 messages using the called and calling numbers. If the calling/called number of a Virtual Trunk session matches the number specified, then the messages to and from the Virtual Trunk are traced. Where: • calling/called number indicates the telephone number to trace on. The value can be a number from 1 to 32 digits and can be a partial calling/called number. • MsgRecv specifies if the messages sent to the specified channel should be traced. The values are ON or OFF. • MsgSend specifies if the messages sent from the specified channel should be traced. The values are ON or OFF.

CLI Command	Description
H323CallTrace ch <beginning channel #> <ending channel #> <MsgRecv> <MsgSend>	Traces a range of Virtual Trunk channels. Where: • beginning channel # indicates the channel number to trace. The values range from 0 - maximum channel number. • ending channel # indicates the channel number to trace. The values range from 0 - maximum channel number, but must be greater than the beginning channel number. • MsgRecv specifies if the messages sent to the designated channel should be traced. The values are ON or OFF. • MsgSend specifies if the messages sent from the designated channel should be traced. The values are ON or OFF. ***Note:*** Replaces the H323CallTrace <beginning channel #> <ending channel #> <MsgRecv> <MsgSend> command.
H323CallTrace num <calling/called number> <NPI> <TON> <MsgRecv> <MsgSend>	Traces H.323 messages using the calling or called number. If the calling/called number of a Virtual Trunk session matches the number specified, and the specified NPI and TON values match the call type, then the messages to and from the Virtual Trunk are traced. Where: • calling/called number indicates the telephone number to trace on. The value can be a number from 1 to 32 digits. • NPI specifies the numbering plan identifier for which calls using this numbering plan are to be traced. The values are: 0 - ALL 1 - Unknown 2 - ISDN/telephony numbering plan (E.164) 4 - E.163 5 - Telex numbering plan (F.69) 6 - Data numbering plan 7 - National standard numbering plan • TON specifies the type of number to use as a filter for tracing. Only calls using this TON setting are traced. Values are: 0 - ALL 1 - Unknown Number 2 - International Number 3 - National Number

CLI Command	Description
	4 - Network Specific Number 5 - Subscriber Number 6 - Level 1 Regional 7 - Level 0 Interface • MsgRecv specifies if the messages sent to the designated channel should be traced. The values are ON or OFF. • MsgSend specifies if the messages sent from the designated channel should be traced. The values are ON or OFF.
help H323CallTrace	Describes the H323CallTrace commands. It supplies each command's parameters and possible parameter values. The command also supplies a list of associated CLIs (that is, H323TraceShow and H3232Output).
H323Output <Output_Destination> <File Pathname>	Specifies where the output for the trace tool is to be directed. Where: • Output_Destination specifies where all the trace messages for H323CallTrace are to be directed.The values are: 1 = TTY 2 = RPTLOG 3 = File 4 = File and TTY • File Pathname specifies the file to output to if option 3 or 4 is selected.
H323TraceShow	Displays the trace settings, including the output destination and filename, as well as all active traces for the H323CallTrace trace tool.

Network Connection Service trace tool commands

Table 23 "NCS CLI commands" (page 312) includes the protocol trace tool CLI commands for the Network Connection Service (NCS) applicable to the Signaling Server and the Voice Gateway Media Cards. They are issued from the OAM shell.

Table 23
NCS CLI commands

CLI command	Description
tpsARTrace IP <IP address>	Allows tracing of the tpsAR protocol, which is used to determine where an IP Phone should register.

CLI command	Description
ID <user ID> ALL	Where: • IP address - a string containing the IP Phone’s IP address • user UID - the ID of the IP Phone to be traced (the DN used to log in) or the H323_Alias of where the IP Phone is trying to register • ALL - all IP Phones are to be monitored
tpsARTraceOff IP <IP address> ID <user ID> ALL	Removes the specified endpoint from the list of endpoints to be traced.
tpsARTraceAllOff	Turns off the trace for all tpsAR trace identifiers.
tpsAROutput <Output_Destination> <"File Pathname"	Sets the output for all tpsAR protocol traces. Where: • Output_Destination specifies where all the trace messages for the tpsARTraceSet are to be directed and whether the command is run from the Voice Gateway Media Card or the vxWorks shell prompt. The values are: 1 = TTY 2 = RPTLOG 3 = File 4 =TTY + File If the command is run from the OAM prompt or PDT prompt on the Signaling Server, then the values are the actual word, not a number: TTY RPTLOG FILE TTY+FILE • "File Pathname" is a string encapsulated in quotes. It specifies the file to output to if option 3 or 4 was selected.

CLI command	Description
tpsARTraceSettings	Displays the trace tool settings, which endpoints are being traced, and where the trace output is being directed.
tpsARTraceHelp	Displays a list of all CLIs used for tracing tpsAR protocol messages, including usage and parameters.

NRS database CLI commands

Table 24 "NRS database CLI commands — OAM shell" (page 314) includes the NRS CLI commands applicable to the Signaling Server. They are issued from the OAM shell.

Table 24
NRS database CLI commands OAM shell

CLI command	Description
nrsGWEndpointShow	Lists all the NRS endpoints with corresponding IP addresses.
nrsUserEPShow	Lists all the NRS users with corresponding IP addresses.
nrsCollaboratingServerShow	Lists all the Collaborating Servers in the database.
nrsL0DomainShow	Lists all the Level 0 regional domains.
nrsL1DomainShow	Lists all the Level 1 regional domains.
nrsRoutingEntryShow	Lists all the Routing Entries in the database.
nrsServiceDomainsShow	Lists all the service provider domains.
nrsGWEndpointQuery	Queries an NRS endpoint with IP and protocol information.
nrsUserEPQuery	Queries an NRS endpoint with IP and protocol information.
nrsL0DomainQuery	Queries a Level 0 regional domain with E164 information.
nrsL1DomainQuery	Queries a Level 1 regional domain.
nrsServiceDomainQuery	Queries a service provider domain.
nrsCollaboratingServerQuery	Queries one Collaborating Server from the database.
nrsDefaultRouteQuery	Queries an NRS default route.
nrsDBShow	DIsplays the state of the Primary and Alternate NRS database and the local NRS database.
NrsOmmShow	Shows the SIP and H.323 NRS statistics for the current hour.
NrsOmmAvShow	Shows the SIP and H.323 NRS total statistics and average statistics for the last seven days.

Table 25 "NRS database CLI commands — PDT shell" (page 315) includes the NRS CLI commands applicable to the Signaling Server. These commands are applicable to the database. They are issued from the PDT shell.

Table 25
NRS database CLI commands PDT shell

CLI command	Description
nrsDbCutover	Switches the active and standby database access pointer.
nrsDbCommit	Mirrors data from active schema to standby schema.
nrsDbCommitNow	Performs cutover and commit in one command.
nrsDbRollback	Undoes the changes.
nrsDbRevert	After the cutover, this command switches the active and standby database access pointer back.
disNRS	Gracefully disables the NRS server service. ***Note:*** This command should not interrupt the existing calls.
forcedisNRS	Forces the NRS server out-of-service.
enlNRS	Enables the SIP Redirect Server service.
nrsSIPTestQuery	Queries a SIP Routing Entry with DN and cost information.
nrsGKTestQuery	Queries an H.323 Routing Entry with DN and cost information.

Stand-alone NRS CLI commands

Table 24 "NRS database CLI commands — OAM shell" (page 314) lists CLI commands for an NRS running on a stand-alone Signaling Server. They are issued from the PDT shell.

Table 26
Stand-alone NRS CLI commands

CLI command	Description
adminUserPasswordChange [userID]	
	Changes the administrator-level user password for an NRS running on a stand-alone Signaling Server, where: • userID = userID of administrator-level user
adminUserCreate [userID]	Creates an administrator-level user of an NRS running on a stand-alone Signaling Server, where: • userID = userID of new administrator-level user

CLI command	Description
adminUserDelete [userID]	Deletes an administrator-level user of an NRS running on a stand-alone Signaling Server, where: • userID = userID of administrator-level user to be deleted
adminAccountShow	Displays the userID and access privileges for all users of an NRS running on a stand-alone Signaling Server.

ISDN to and from SIP mapping CLI commands

Table 27 "ISDN-to-SIP commands" (page 316) shows the commands for mapping from ISDN to SIP, and Table 28 "SIP-to-ISDN commands" (page 316) shows the commands for mapping from SIP to ISDN. These commands are issued from the PDT prompt.

Table 27
ISDN-to-SIP commands

Command	Description
isdn2SipSet num1, num2	Changes the ISDN cause code to the SIP status code mapping. Where: • num1 is the ISDN cause code • num2 is the SIP status code
isdn2SipReset num	Resets a single ISDN cause code to the default SIP status code mapping. Where num is the ISDN cause code.
isdn2SipResetAll	Resets all the ISDN cause codes to the default SIP status code mappings.
isdn2SipShow num	Shows one specific ISDN cause code to SIP status code mapping.
isdn2SipShowAll	Shows all mappings from ISDN cause codes to SIP status codes.

Table 28
SIP-to-ISDN commands

Command	Description
sip2IsdnSet num1, num2	Changes the SIP status code to the ISDN cause code mapping. Where:

Command	Description
	• num1 is the SIP status code • num2 is the ISDN cause code
sip2IsdnReset num	Resets a single SIP status code to the default ISDN cause code mapping. Where num is the SIP status code.
sip2IsdnResetAll	Resets all SIP status codes to the default ISDN cause code mappings.
sip2IsdnShow num	Shows one specific SIP status code to ISDN cause code mapping. Where num is the ISDN cause code.
sip2IsdnShowAll	Shows all mappings from SIP status code to ISDN cause code.

Call Server commands

Manage Virtual Trunk route members

Use the commands in LD 32 to enable or disable Virtual Trunk route members, or to display information about route members.

LD 32 Manage Virtual Route members.

Command	Description
DIS VTRM <cust #> <route #>	Disables all route members in a customer’s route. This command: • disconnects all active calls associated with the trunks • disables all route members on the Call Server • unregisters all trunks • removes them from the RLM table On the Signaling Server, all trunks are removed from the Signaling Server list.

Command	Description
ENL VTRM <cust #> <route #>	Enables all the route members (Virtual Trunks) This command: • enables all route members in a customer’s route • enables all route members • register the member • puts the members into the RLM table On the Signaling Server, all trunks are put on the Signaling Server list.
STAT VTRM <cust#> <rout#> start_mb# end_mb#	Displays the Virtual Trunk status specified by customer number, route number, and starting and ending member number. ***Note:*** Also see "LD 32 — STAT VTRM commands" (page 319) for additional usage of the STAT VTRM command.

Status commands

Use the STAT LINK and STAT SERV commands in LD 117 and the STAT VTRM command in LD 32 to display link information of connected services.

LD 117 STAT LINK and STAT SERV commands

Command	Description
stat link ip <IP address>	Displays the link information and link status of the server with the specified IP address or contained specified subnet.
stat link srv ss	Displays the link information and link status of the Signaling Servers.
stat link name <hostname>	Displays the link information and link status of the server with the specified hostname.
stat link node <node ID>	Displays the link information and link status of the server with the specified node ID.
stat serv ip <IP address>	Displays the information of the server with the specified IP address or contained specified subnet.

Command	Description
stat serv app <applicationType>	Displays the information of the server running the specified application. Where application type can be: • LTPS (Line TPS) • VTRK (Virtual Trunk) • GK (Gatekeeper)
stat serv node <node ID>	Displays the information of the server with the specified node ID.
stip tn <tn>	Displays the IP information and status of the specified TN.
stip type ipti	Displays the IP information and status of all TNs that are of IPTI (Virtual Trunk and ITG Trunk) type.

LD 32 STAT VTRM commands

STAT command	Description
STAT VTRM	Displays a status summary for all IP Peer Virtual Trunk routes associated with all customer numbers.
STAT VTRM <Cust>	Displays a status summary for all IP Peer Virtual Trunk routes associated with the customer number.
STAT VTRM <Cust> <Rout>	Displays a status summary for the specified IP Peer Virtual Trunk route.
STAT VTRM <Cust> <Rout> <Starting Member> <number of trunks>	Displays a status summary for the specified IP Peer Virtual Trunk route followed by a listing of Virtual Trunk TNs in the specified range.
STAT VTRM <Cust> SIP / H323	Displays a status summary for all IP Peer Virtual Trunk routes of the specified VoIP signaling protocol associated with the customer number.
STAT VTRM <Cust> <Rout> ALL	Displays a status summary for the specified IP Peer Virtual Trunk route followed by a listing of all Virtual Trunk TNs in the specified route.
STAT VTRM <Cust> <Rout> REG	Displays a status summary for the specified IP Peer Virtual Trunk route followed by a listing of registered Virtual Trunk TNs in the specified route.
STAT VTRM <Cust> <Rout> UNR	Displays a status summary for the specified IP Peer Virtual Trunk route followed by a listing of unregistered Virtual Trunk TNs in the specified route.
STAT VTRM <Cust> <Rout> BUSY	Displays a status summary for the specified IP Peer Virtual Trunk route followed by a listing of busy Virtual Trunk TNs in the specified route.

STAT command	Description
STAT VTRM <Cust> <Rout> IDLE	Displays a status summary for the specified IP Peer Virtual Trunk route followed by a listing of idle Virtual Trunk TNs in the specified route.
STAT VTRM <Cust> <Rout> MBSY	Displays a status summary for the specified IP Peer Virtual Trunk route followed by a listing of maintenance busy Virtual Trunk TNs in the specified route.
STAT VTRM <Cust> <Rout> DSBL	Displays a status summary for the specified IP Peer Virtual Trunk route followed by a listing of disabled Virtual Trunk TNs in the specified route.
STAT VTRM <Cust> <Rout> LCKO	Displays a status summary for the specified IP Peer Virtual Trunk route followed by a listing of locked out Virtual Trunk TNs in the specified route.
ENL VTRM <Cust> <Rout>	Enables all IP Peer Virtual Trunk TNs in the specified route associated with the specified customer.
DIS VTRM <Cust> <Rout>	Disables all IP Peer Virtual Trunk TNs in the specified route associated with the specified customer.

Signaling Server error logging and SNMP alarms

SNMP alarms

When the IP Peer Gateway and NRS applications generate alarms, these alarms are output from the Signaling Server. For example, an SNMP alarm is generated if the Signaling Server loses the link to the Call Server.

When an error or specific event occurs, the Signaling Server sends an alarm trap to any configured trap destinations. TM receives SNMP traps from the CS 1000 Systems and stores the traps in a circular log file on the TM Server. The TM Alarm Notification application monitors incoming traps and notifies the appropriate users of important events and alarms. For more information about TM alarm management, refer to *Communication Server 1000E: Maintenance* (NN43041-700).

HPOpenView is an example of an SNMP manager.

For detailed information, refer to *Simple Network Management Protocol: Description and Maintenance* (NN43001-719).

Error logging

An SNMP alarm places a system error message into the Signaling Server’s error log file. The error log file can be viewed using Element Manager. The file can also be viewed in any text browser once the file is uploaded to an FTP host using the LogFilePut command.

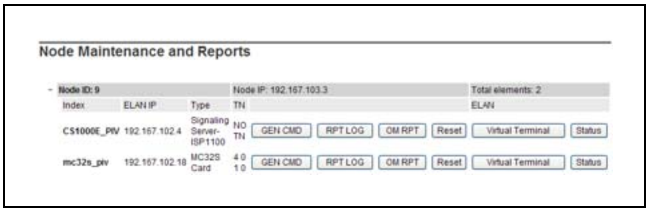
Use following procedure to view the error log in EM Navigator.

Procedure 30
Viewing the error log file

Step	Action
1	Select **Tools > Logs and Reports > IP Telephony Nodes** from the EM Navigator.

The **Node Maintenance and Reports** web age opens, as shown in Figure 169 "Node Maintenance and Reports web page" (page 321).

Figure 169
Node Maintenance and Reports web page



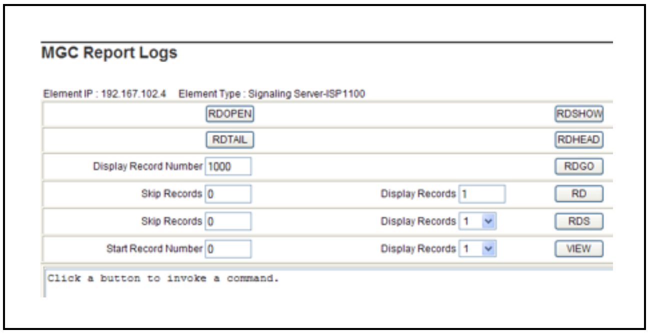
Node Maintenance and Reports

- Node ID: 9			Node IP: 192.167.103.3	Total elements: 2
Index	ELAN IP	Type	TN	ELAN
CS1000E_PIV	192.167.102.4	Signaling Server-ISP1100	NO TN	
mc32s_piv	192.167.102.18	MC32S Card	4 0 1 0	

2 Click **RPT LOG** for the **Index** entry containing the associated Signaling Server.

The **MGC Report Logs** web age opens, as shown in Figure 170 "MGC Report Logs web page" (page 321). For more information about this page, refer to *Element Manager: System Administration* (NN43001-632).

Figure 170
MGC Report Logs web page



MGC Report Logs

Element IP : 192.167.102.4 Element Type : Signaling Server-ISP1100

Display Record Number 1000

Skip Records 0 Display Records 1

Skip Records 0 Display Records 1

Start Record Number 0 Display Records 1

```
Click a button to invoke a command.
```

3 Click a button to invoke a command.

—End—

The **Node Maintenance and Reports** web page provides status information about the system and access to diagnostic tools. These tools enable users to issue commands to maintain CS 1000E and CS 1000M components. Use features on the **Node Maintenance and Reports** web page to perform maintenance tasks, troubleshooting, and problem resolution.

The **System Status** web page provides status information about the system and access to diagnostic tools. These tools enable users to issue commands to maintain CS 1000E and CS 1000M components. Use features on the **System Status** web page to perform maintenance tasks, troubleshooting, and problem resolution.

Error message format

ITG messages are generated from the Voice Gateway Media Cards and the Signaling Server. ITS messages are generated from the IP Phone and are reported through the Signaling Server.

The format of the ITG and ITS error messages is ITGsxxx or ITSsxxx, where sxxx is a four digit number. For example, ITG0351.

The first digit of the four digit number in the error message represents the severity category of the message. The severity categories are:

1 = Critical

2 = Major

3 = Minor

4 = Warning

5 = Cleared (Info)

6 = Indeterminate (Info)

> ***Note:*** Message numbers beginning with 0 do not follow this format.

For a detailed list of the ITG and ITS error messages, refer to *Software Input/Output: System Messages* (NN43001-712).

Appendix A
ISDN/H.323 mapping tables

Nortel proprietary Private UDP numbers (ESN LOC) are encoded as Private Level 1 Regional numbers in H.323. CDP numbers are encoded as Private Level 0 Regional numbers in H.323. In H.225.0 (Q.931) messages, public numbers (E.164) are encoded in the Information Element (IE). Private numbers are encoded in the User to User Information Element (UUIE). On reception, both the IE and UUIE are accepted. If both are included, preference is given to the proper format (that is, the IE for public numbers and the UUIE for private numbers). The numbers in the Signaling Server are encoded using the Universal ISDN Protocol Engine (UIPE) format (which is different from Q.931/MCDN/H.323). Table 29 "Mapping from UIPE to H.225.0 for NPI" (page 323) to Table 38 "Mapping from H.225.0 UUIE to UIPE for Unqualified Number" (page 326) describe the mapping.

Table 29
Mapping from UIPE to H.225.0 for NPI

Numbering Plan Indicator (NPI)	UIPE	H.225.0 IE NPI	H.225.0 UUIE NPI
Unknown	0000 (0)	1001 (9)	privateNumber
ISDN/Telephony (E.164)	0001 (1)	0001 (1)	publicNumber
Private	0010 (2)	1001 (9)	privateNumber
Telephony (E.163)	0011 (3)	0001 (1)	publicNumber
Telex (F.69)	0100 (4)	0100 (4)	N/A

Numbering Plan Indicator (NPI)	UIPE	H.225.0 IE NPI	H.225.0 UUIE NPI
Data (X.121)	0101 (5)	0011 (3)	N/A
National Standard	0110 (6)	1000 (8)	N/A

Table 30
Mapping from UIPE to H.225.0 for TON (NPI = E.164/E.163)

TON (NPI=E.164/E.163)	UIPE TON	H.225.0 IE TON	H.225.0 UUIE TON
Unknown	000 (0)	000 (0)	unknown
International number	001 (1)	001 (1)	internationalNumber
National number	010 (2)	010 (2)	nationalNumber
Special number	011 (3)	011 (3)	networkSpecificNumber
Subscriber number	100 (4)	100 (4)	subscriberNumber

Table 31
Mapping from UIPE to H.225.0 for TON (NPI = Private)

TON (NPI = Private)	UIPE TON	H.225.0 IE TON	H.225.0 UUIE TON
Unknown	000 (0)	000 (0)	unknown
ESN LOC (UDP)	101 (5)	000 (0)	level1RegionalNumber
ESN CDP	110 (6)	000 (0)	localNumber
ESN Special Number	011 (3)	000 (0)	pISNSpecificNumber
Note: When NPI = Private, the number digits are encoded in the privateNumber of PartyNumber, which includes the Type of Number (TON). The TON in the H.225.0 IE are ignored on receipt and coded as Unknown (that is, 0000.) In H.323 version 4.0, *publicNumber* is renamed *e164Number*			

Table 32
Mapping from H.225.0 Information Element to UIPE for NPI

NPI	H.225.0 IE NPI	UIPE NPI
ISDN/Telephony (E.164)	0001 (1)	0001(1)
Private	1001 (9)	0010 (2)
Telephony (E.163)	0010 (2)	0011 (3)
Telex (F.69)	0100 (4)	0100 (4)
Data (X.121)	0011 (3)	0101 (5)

NPI	H.225.0 IE NPI	UIPE NPI
National Standard	1000 (8)	0110 (6)
Unknown	all others	0000 (0)

Table 33
Mapping from H.225.0 Information Element to UIPE for TON (NPI = E.164/E.163)

TON (NPI = E.164/E.163)	H.225.0 IE TON	UIPE TON
International number	001 (1)	001 (1)
National number	010 (2)	010 (2)
Network specific number	011 (3)	011 (3)
Subscriber number	100 (4)	100 (4)
Unknown	all others	000 (0)

Table 34
Mapping from H.225.0 Information Element to UIPE for TON (NPI = Private)

TON (NPI = Private)	H.225.0 IE TON	UIPE TON
Level 1 Regional Number	010 (2)	101 (5)
Local Number/ Level 0 Regional	100 (4)	110 (6)
PISN Specific Number	011 (3)	011 (3)
Unknown	all others	000 (0)

Note: When NPI = Private, precedence is given to any number in the H.225.0 UUIE. The H.225.0 IE is only used if the H.225.0 UUIE is not present. The Presentation Indicator and Screening Indicator are always in the information H.225.0 IE. The H.225.0 UUIE is only used if the H.225.0 IE is not present. In H.323 version 4.0, *publicNumber* is renamed *e164Number*.

Table 35
Mapping from H.225.0 UUIE to UIPE for NPI

NPI	H.225.0 UUIE NPI	UIPE NPI
ISDN/Telephony (E.164)	publicNumber	0001 (1)
Private	privateNumber	0010 (2)

Table 36
Mapping from H.225.0 UUIE to UIPE for TON (NPI = E.164/E.163)

TON (NPI = E.164/E.163)	H.225.0 UUIE TON	UIPE TON
International number	internationalNumber	001 (1)
National number	nationalNumber	010 (2)

TON (NPI = E.164/E.163)	H.225.0 UUIE TON	UIPE TON
Network specific number	networkSpecificNumber	011 (3)
Subscriber number	subscriberNumber	100 (4)
Unknown	all others	000 (0)

Table 37
Mapping from H.225.0 UUIE to UIPE for TON (NPI = Private)

TON (NPI = Private)	H.225.0 UUIE TON	UIPE TON
Level 1 Regional Number	level1 RegionalNumber	101 (5)
Local Number/ Level 0 Regional	localNumber	110 (6)
PISN Specific Number	pISNSpecificNumber	011 (3)
Unknown	all others	000 (0)

Table 38
Mapping from H.225.0 UUIE to UIPE for Unqualified Number

Unqualified Number	H.225.0 UUIE	UIPE NPI	UIPE TON
Dialed Digits	e164	0000 (0)	0000 (0)
Note: In H.323 version 4.0, *e164* is renamed *dialedDigits*. In H.323 version 4.0, *publicNumber* is renamed *e164Number*.			

Appendix B
H.323 Gatekeeper overlap signaling support

Contents

This section contains information on the following topics:

Overlap signaling and H.323 Gatekeeper-routed calls

With H.323 Gatekeeper-routed signaling, admission messages are exchanged between the endpoints and the H.323 Gatekeeper on RAS channels. The H.323 Gatekeeper receives the call-signaling messages on the call-signaling channel from one endpoint and routes them to the other endpoint on the call-signaling channel of the other endpoint.

With direct-routed signaling in the admission confirmation, the H.323 Gatekeeper indicates that the endpoints can exchange call-signaling messages directly. The endpoints exchange the call signaling on the call-signaling channel.

If the H.323 Gatekeeper uses H.323 Gatekeeper routing, it may or may not also use "pre-granted admission". That is, it may not (and usually does not) need the Admission Request message. As a result, the SETUP message is sent to the H.323 Gatekeeper by the Gateway, and all further processing is done by the H.32 gatekeeper.

For processing to succeed, the H.323 Gatekeeper must be fully compliant with H.323 overlap signaling. That is, the H.323 Gatekeeper must be able to receive multiple messages with digits — the SETUP and subsequent INFORMATION messages.

When the calls are H.323 Gatekeeper-routed, the H.323 Gatekeeper must have the ability to do the following:

- decode digits from SETUP and INFORMATION messages
- perform the address resolution

It must then originate overlap calls (or overlap to en bloc, if necessary) to the destination.

Mixed networks of overlap and en bloc H.323 Gatekeepers

Overlap-capable H.323 Gatekeepers can co-reside with H.323 Gatekeepers that cannot do all the necessary overlap functions. As a result, the H.323 Gatekeepers and the H.323 Gateways must be able to accommodate this occurrence.

The simplest example is in the Location Request (LRQ) handling.

- H.323 versions prior to H.323 Release 4.0 do not support the "incomplete address" reason code in the Location Reject (LRJ). As a result, if an overlap H.323 Gatekeeper is registered to a local overlap-capable H.323 Gatekeeper, then the H.323 Gatekeeper sends a digit string in the ARQ that the H.323 Gatekeeper cannot resolve, and that H.323 Gatekeeper queries its peers.
- However, if the remote H.323 Gatekeeper supports H.323 Release 3 or earlier (or does not support overlap signaling even though the H.323 Gatekeeper conforms to Release 4), no "incomplete" message can be returned.

Any LRQ sent to the remote H.323 Gatekeeper is either rejected with a cause indicating failure, or is ignored. The local H.323 Gatekeeper can determine its own capabilities; however, it cannot determine the capabilities of the remote H.323 Gatekeeper. The local H.323 Gatekeeper also cannot "guess" the returned reasons. For example, a "request denied" may have been triggered by a messaging error or by the sender not having any way to indicate an incomplete number.

To resolve this issue, when a local H.323 Gatekeeper determines from the local provisioning and received responses that no completion can occur, it returns either the Default Route as the destination in the ACF or an ARJ indicating failure to the gateway. However, because differentiation between a "normal ACF" and a "default route ACF" cannot be made, the non-standard data is enhanced to indicate this to the gateway. This indication is done in the non-standard data because the element includes vendor information and, as a result, non-Nortel gateways can read the manufacturer information and ignore the data.

As part of the protocol, all endpoints supporting the protocol must have a predefined way to handle the indication. That is, if the H.323 Gatekeeper indicates that a default route was selected (or would have been selected if the entry had been provisioned) by sending the Default Route Indicator (DRI), then any gateway supporting the protocol must have a predetermined general handling procedure to handle the indication.

The rationale for the general handling procedure is simple. The protocol is designed to be fully forward-compatible. If any recommendation (DRI Recommendation [DRIR]) sent to a gateway by the H.323 Gatekeeper cannot be found in the list of DRIR values understood by the gateway, then the gateway must have a defined procedure for handling this event. That is, the following algorithm applies:

- If the gateway recognizes the recommendation and it is completely valid, the gateway uses the recommendation.
- If the gateway recognizes the recommendation but there is a reason that it cannot apply (for example, if a recommendation such as "wait for more digits" existed but the call was from an en bloc gateway and there are no more digits), the gateway uses either the general handling procedure or some other selected procedure.
- If the gateway does not recognize the recommendation, it uses the general handling procedure. This includes recommendations that have not yet been defined, so the protocol covers forward compatibility.

The importance of this capability within a mixed network is simple. If LRQs are broadcast to the peer H.323 Gatekeepers and no positive responses return, then this may be because no positive responses are possible; the number may be completely undefined. On the other hand, the H.323 Gatekeeper "may just not have responded" but the number may be valid.

H.323 Gatekeeper recommendations for overlap signaling in mixed overlap and en bloc networks

There are two key concepts behind the recommendations:

- First, calls placed using overlap signaling to an en bloc gateway use processing resources that they do not need to use. The SETUP and subsequent INFORMATION messages can trigger multiple Admission Requests to the H.323 Gatekeeper, which in turn can trigger Location Requests throughout a significant part of the IP telephony network. If the average count of ARQs for each call doubles, then the maximum through-put of the H.323 Gatekeeper in calls per hour is halved. For an en bloc call, there is only a single ARQ, which either succeeds or fails the first time.
- Second, calls placed to an en bloc gateway can terminate prematurely to a terminal to handle failed calls. That is, misdialed numbers can route

to a specified answering position such as an Attendant, the Security Desk, or some other site. If the destination gateway is provisioned with this sort of capability, then the calls that should have been rejected and sent back to the originator for overlap-to-en bloc conversion. However, the calls receive manual overlap-to-en bloc conversion, as the caller tells the party (answering the intercepted call) the destination that the caller really wanted.

A third item acting as a base for the recommendations is call control traffic on the Signaling Server. Although the call control traffic is not heavy enough to make optimization necessary, it does provide additional justification. Overlap signaling adds some overhead, but much less on the H.323 signaling gateway than on the H.323 Gatekeeper.

With this background information, the following recommendations apply:

1. Even though a gateway may support overlap signaling, if the H.323 Gatekeeper that the gateway uses does not support overlap signaling, then do not provision the gateway as overlap.

   Note: All further recommendations assume that the H.323 Gatekeeper supports overlap signaling.

2. Assume that an en bloc destination is registered with the local H.323 Gatekeeper. If this destination is known to be en bloc only, but returns the "unassigned number" or "invalid number format" cause codes, then the administrator can provision the originating Call Server to leave this call as an overlap call. The returned cause code in the RELEASE COMPLETE message triggers overlap-to-en bloc conversion. However, the Overlap Length (OVLL) prompt (in LD 86) must be configured to a value that gives a reasonable probability that the H.323 Gatekeeper can resolve the call on the first attempt. This is to avoid excessive querying of the H.323 Gatekeeper.
3. Assume that an en bloc destination is registered with the local H.323 Gatekeeper. If this destination is known to be en bloc only, but either will not return the desired cause code or will intercept the call, then provision the entry on the originating Call Server with an en bloc Route List Index (RLI). That is, even though the D-channel can accept overlap signaling, define the RLI used for this call with an OVLL of 0. This forces the call into en bloc handling.
4. Assume that an en bloc destination is registered with a remote H.323 Gatekeeper. En bloc destinations, that must be reached using Location Request (LRQ) messages to their H.323 Gatekeeper, are subject to the limitations of that H.323 Gatekeeper. The en bloc destination is also subject to their own limitations regarding handling incomplete numbers. If possible, these destinations should be provisioned as en bloc using

OVLL 0, since the remote H.323 Gatekeeper may not be able to handle an overlap call LRQ with an incomplete called-party digit string.

5. Assume that an en bloc destination is registered with a remote H.323 Gatekeeper and OVLL on the originating Call Server is not configured as 0. H.323 Gatekeepers that cannot support overlap signaling may not be able to respond to an LRQ message with an LRJ message to reject the call. If this occurs, the Signaling Server attempts overlap-to-en bloc conversion (unless a prior reply indicated either a successful termination at another destination or that the number was incomplete on another H.323 Gatekeeper). If the local H.323 Gatekeeper fails to receive any response to its LRQ from one or more H.323 Gatekeepers while all others indicate failure, and it has a default route defined, this is provided to the gateway. In addition, the H,323 Gatekeeper provides an indication that the call was terminated to the default route. This allows the gateway to either route the call to the default route destination, or to try overlap-to-en bloc conversion. Therefore, Nortel recommends that the administrator provision any CS 1000 Release 4.0 (or later) H.323 Gatekeepers (with the NRS) with a default route.

6. Assume that a destination is known to be en bloc and OVLL is configured to 0. For all these en bloc destination numbers, if the length of the number is known, then ensure that the Flexible Length (FLEN) prompt is provisioned for that number. Provisioning the FLEN of an eight-digit number as 8 triggers an immediate SETUP on dialing the eighth digit. If the FLEN is longer (or configured to 0), then the Call Server runs an end-of-dial timer to determine whether the number is complete. Failing to configure the FLEN correctly adds several seconds to the post-dial delay (the time between the last digit being dialed and hearing ringback) for the call.

7. If a destination is known to be overlap-capable, the best performance is possible by using overlap dialing. This allows the H.323 Gatekeepers to minimize their database size. A ’smaller’ database speeds up responses to queries and allows calls to reach the destination faster. Also, when a call tandems to an overlap-capable PSTN, this gives the best end-to-end performance. So, for destinations in overlap-capable countries, it is a good rule of thumb to always provision any overlap-capable destination to use overlap signaling.

8. If a network is located in an en bloc-only jurisdiction, then there is no harm in provisioning the gateways that can do overlap dialing to receive overlap calls. In this manner, if a new domain from an overlap-compatible area is added later, then all calls that are received as overlap (from the new domain) can be processed more efficiently.

9. If the call terminates on an en bloc-only PSTN, do not use overlap for this call. As an example, the North American dialing plan uses an NPA-NXX-XXXX format. North America also uses en bloc to the PSTN.

Therefore, for calls provisioned on the originating Call Server as NPA and NXX calls, do not use overlap. These calls must use OVLL 0 RLIs.

10. If a call is a remote E.164 plan (International, National, or Local/Subscriber) type of number, then this call must traverse the IP network as a Special Number (SPN). Otherwise, all overlap capability is lost. At the node where the call tandems to the PSTN, the type of number is changed to International, National, or Local (as applicable). However, if the PSTN supports these as overlap calls, then it is guaranteed that the node must be able to receive them as overlap as well. Therefore, provision the originating Call Server with this call as an SPN, and prefix any local numbers with the national code. Then, if it is required that local calls not have national prefixes at the destination, then when the call to the PSTN breaks out to the PSTN, remove any national prefixes from calls going to the local area.

Appendix C
ISDN cause code to SIP status code mapping tables

When an *ISDN: Release* message is received before receiving a SIP final response, a 4xx/5xx message is sent to the far end indicating a corresponding error situation. Table 39 "ISDN cause code to SIP status code mapping" (page 334) maps the cause code in the *ISDN: Release* message to SIP status code according to RFC 3398. If an ISDN cause value other than those listed in Table 39 "ISDN cause code to SIP status code mapping" (page 334) is received, the default SIP response *500 Server internal error*is used. If a SIP status code other than those listed is received, the default ISDN cause code is *21 call rejected*.

Note that the SIP code to ISDN cause code is not one-to-one mapping. Several cause codes can map to one single SIP response. For example, ISDN reason 1, 2, and 3 map to SIP *404* message, but the SIP *404* message only maps to ISDN reason 1. This implies that, when mapping a 4xx/5xx message to ISDN cause value, some information may be lost and further investigation should be done on an individual call basis.

The SIP warning phrase is modified to include the ISDN cause code. For example, *503 Service unavailable ISDN: 34*. With MCDN tunneling, the ISDN cause code is presented in tunneled MCDN message as well as the SIP message. The receiver of such a message uses the cause code in MCDN message instead of the SIP warning phrase.

Table 39 "ISDN cause code to SIP status code mapping" (page 334) shows the ISDN cause code to SIP status code mapping, and Table 40 "ISDN cause code to SIP status code mapping" (page 335) shows the SIP error response to ISDN cause code mapping.

> ***Note:*** If desired, a user can change those default mappings through CLI commands.

Table 39 "ISDN cause code to SIP status code mapping" (page 334) shows the ISDN cause code to SIP status code mapping.

Table 39
ISDN cause code to SIP status code mapping

ISDN cause code	SIP response
1 unallocated number	404 Not Found
2 no route to network	404 Not Found
3 no route to destination	404 Not found
16 normal call clearing	BYE or Cancel
17 user busy	486 Busy here
18 no user responding	408 Request Timeout
19 no answer from the user	480 Temporarily unavailable
20 subscriber absent	480 Temporarily unavailable
21 call rejected	403 Forbidden (If the cause location is ’user’, then code 603 could be given rather than the 403 code)
22 number changed (w/o diagnostic)	410 Gone
22 number changed (w/ diagnostic)	301 Moved Permanently
23 redirection to new destination	410 Gone
26 non-selected user clearing	404 Not Found
27 destination out of order	502 Bad Gateway
28 address incomplete	484 Address incomplete
29 facility rejected	501 Not implemented
31 normal unspecified	480 Temporarily unavailable
34 no circuit available	503 Service unavailable
38 network out of order	503 Service unavailable
41 temporary failure	503 Service unavailable
42 switching equipment congestion	503 Service unavailable
47 resource unavailable	503 Service unavailable
55 incoming calls barred within CUG	403 Forbidden
57 bearer capability not authorized	403 Forbidden
58 bearer capability not presently available	503 Service unavailable
65 bearer capability not implemented	488 Not Acceptable Here
70 only restricted digital avail	488 Not Acceptable Here
79 service or option not implemented	501 Not implemented

ISDN cause code	SIP response
87 user not member of CUG	403 Forbidden
88 incompatible destination	503 Service unavailable
102 recovery of timer expiry	504 Gateway timeout
111 protocol error	500 Server internal error
127 interworking unspecified	500 Server internal error

Table 40 "ISDN cause code to SIP status code mapping" (page 335) shows the SIP error response to ISDN cause code mapping.

Table 40
ISDN cause code to SIP status code mapping

SIP response	ISDN cause code
400 Bad Request	41 Temporary Failure
401 Unauthorized	21 Call rejected
402 Payment required	21 Call rejected
403 Forbidden	21 Call rejected
404 Not found	1 Unallocated number
405 Method not allowed	63 Service or option unavailable
406 Not acceptable	79 Service/option not implemented
407 Proxy authentication required	21 Call rejected
408 Request timeout	102 Recovery on timer expiry
410 Gone	22 Number changed (without diagnostic)
413 Request Entity too long	127 Interworking
414 Request-URI too long	127 Interworking
415 Unsupported media type	79 Service/option not implemented
416 Unsupported URI Scheme	127 Interworking
420 Bad extension	127 Interworking
421 Extension Required	127 Interworking
423 Interval Too Brief	127 Interworking
480 Temporarily unavailable	18 No user responding
481 Call/Transaction Does not Exist	41 Temporary Failure
482 Loop Detected	25 Exchange - routing error
483 Too many hops	25 Exchange - routing error
484 Address incomplete	28 Invalid Number Format

SIP response	ISDN cause code
485 Ambiguous	1 Unallocated number
486 Busy here	17 User busy
487 Request Terminated	no mapping
488 Not Acceptable here	by Warning header
500 Server internal error	41 Temporary failure
501 Not implemented	79 Not implemented, unspecified
502 Bad gateway 3	8 Network out of order
503 Service unavailable	41 Temporary failure
504 Server time-out	02 Recovery on timer expiry
505 Version Not Supported	127 Interworking
513 Message Too Large	127 Interworking
600 Busy everywhere	17 User busy
603 Decline	21 Call rejected
604 Does not exist anywhere	1 Unallocated number
606 Not acceptable	by Warning header

Appendix
Passthrough End User License Agreement

WARNING
Do ***not*** contact Red Hat for technical support on your Nortel version of the Linux base operating system. If technical support is required for the Nortel version of the Linux base operating system, contact Nortel technical support through your regular channels.

This section governs the use of the Red Hat Software and any updates to the Red Hat Software, regardless of the delivery mechanism and is governed by the laws of the state of New York in the U.S.A. The Red Hat Software is a collective work under U.S. Copyright Law. Subject to the following terms, Red Hat, Inc. (“Red Hat”) grants to the user (“Customer”) a license to this collective work pursuant to the GNU General Public License. Red Hat Enterprise Linux (the “Red Hat Software”) is a modular operating system consisting of hundreds of software components. The end user license agreement for each component is located in the component’s source code. With the exception of certain image files identified below, the license terms for the components permit Customer to copy, modify, and redistribute the component, in both source code and binary code forms. This agreement does not limit Customer’s rights under, or grant Customer rights that supersede, the license terms of any particular component. The Red Hat Software and each of its components, including the source code, documentation, appearance, structure and organization are owned by Red Hat and others and are protected under copyright and other laws. Title to the Red Hat Software and any component, or to any copy, modification, or merged portion shall remain with the aforementioned, subject to the applicable license. The “Red Hat” trademark and the “Shadowman” logo are registered trademarks of Red Hat in the U.S. and other countries. This agreement does not permit Customer to distribute the Red Hat Software using Red Hat’s trademarks. If Customer makes a commercial redistribution of the Red Hat Software, unless a separate agreement with Red Hat is executed or other permission granted, then Customer must modify any files identified as “REDHAT-LOGOS” and “anaconda-images” to remove all images containing the “Red Hat” trademark or the “Shadowman” logo. As required by U.S. law, Customer represents and warrants that it: (a) understands that the Software is subject to export controls under the U.S. Commerce Department’s Export Administration Regulations (“EAR”); (b) is not located in a prohibited destination country under the EAR or U.S. sanctions regulations (currently Cuba, Iran, Iraq, Libya, North Korea, Sudan and Syria); (c) will not export, re-export, or transfer the Software to any prohibited destination, entity, or individual without the necessary export license(s) or authorizations(s) from the U.S. Government; (d) will not use or transfer the Red Hat Software for use in any sensitive nuclear, chemical or biological weapons, or missile technology end-uses unless authorized by the U.S. Government by regulation or specific license; (e) understands and agrees that if it is in the United States and exports or transfers the Software to eligible end users, it will, as required by EAR Section 740.17(e), submit semi-annual reports to the Commerce Department’s Bureau of Industry & Security (BIS), which include the name and address (including country) of each transferee; and (f) understands that countries other than the United States may restrict the import, use, or export of encryption products and that it shall be solely responsible for compliance with any such import, use, or export restrictions. Red Hat may distribute third party software programs with the Red Hat Software that are not part of the Red Hat Software. These third party programs are subject to their own license terms. The license terms either accompany the programs or can be viewed at http://www.redhat.com/licenses/. If Customer does not agree to abide by the applicable license terms for such programs, then Customer may not install them. If Customer wishes to install the programs on more than one system or transfer the programs to another party, then Customer must contact the licensor of the programs. If any provision of this agreement is held to be unenforceable, that shall not affect the enforceability of the remaining provisions. Copyright © 2003 Red Hat, Inc. All rights reserved. "Red Hat" and the Red Hat "Shadowman" logo are registered trademarks of Red Hat, Inc. "Linux" is a registered trademark of Linus Torvalds. All other trademarks are the property of their respective owners.

Nortel Communication Server 1000

IP Peer Networking Installation and Commissioning

Publication: NN43001-313
Document status: Standard
Document version: 01.01
Document date: 30 May 2007

To provide feedback or report a problem in the document, go to www.nortel.com/documentfeedback.

Sourced in Canada.

The information in this document is subject to change without notice. The statements, configurations, technical data, and recommendations in this document are believed to be accurate and reliable, but are presented without express or implied warranty. Users must take full responsibility for their applications of any products specified in this document. The information in this document is proprietary to Nortel Networks.